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THE BEST OF Audio VOLUME IV

The Cover Equipment: Fuselier 3.8D loudspeaker, Onkyo DX-G10 CD player, and Yamaha DSP-3000 surround processor.

The Cover Photographer: Peter Ardito.

Audio Publishing, Editorial and Advertising Offices, 1515 Broadway, New York, N.Y. 10036.

EDITOR'S OVERVIEW

Dear Reader:

Welcome to Volume IV of *The Best of Audio*. I think that by the time you've turned to this editorial, you will have realized that the magazine's title involves a rather small, second-rate pun. But however good the pun is, we are quite serious about how good the equipment we review here is. And, too, I am quite certain about how good our reviewing technique is. If you are serious about hi-fi gear, then we are the best in the world. We believe, quite literally, that no magazine anywhere publishes more thorough reports, presents more graphs and 'scope photos, tests for more parameters, or includes more data that is the direct result of their own measurement than *Audio*.

In each of the review sections, you will find a selection of test reports which we have performed over the last few years. While the test may not be fresh as today's delivery of milk, the gear tested is—we have checked with the manufacturer to see that no significant parameter has changed since we first published the test. The important thing to remember, however, is that the equipment reviewed here is easily among the finest available. You can rely on its excellence because we have performed exhaustive lab tests and because we have done hours of listening to the reviewed piece in our reference systems.

It's worth mentioning that our equipment reviewers are themselves world-renowned, having made their reputations in their individual fields. All of them are completely familiar with the in's and out's of both practical design and the theory behind the equipment. Some of them actually did engineering production work during their careers, while others have spent decades developing the measurements, either the test procedures or the actual instruments used.

Overall, more than two dozen pieces of gear are covered in this issue of *The Best of Audio*. It is, of course, impossible to cover *all* of the state-of-the-art equipment in such depth in a single magazine; there is, as well, other gear out there that's worthy of your consideration. But whatever you do, don't overlook the components we cover. You may argue, for whatever reason, that something doesn't suit your tastes or isn't practical for your system. Be that as it may, I am certain you'll agree that in terms of sheer performance, what's presented here is truly the best of audio.

Cordially,

Gene

Eugene Pitts
Editor

Audio

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KLIPSCHORN LOUDSPEAKER

Manufacturer's Specifications
System Type: Three-way horn-loaded, for corner placement.
Drivers: 15-inch woofer, 2-inch midrange, and 1-inch tweeter.
Frequency Range: 35 Hz to 17 kHz, ±5 dB.
Sensitivity: 104 dB SPL at 1 meter for 1 watt input.
Crossover Frequencies: 400 Hz and 6 kHz.
Impedance: 8 ohms nominal, 4 ohms minimum.
Recommended Amplifier Power: 20 watts minimum.
Dimensions: 52 in. H x 31 in. W x 29 in. D (132 cm x 79 cm x 74 cm).
Weight: 165 lbs. (74.9 kg).
Price: $1,649 each.
Company Address: Klipsch, P.O. Box 688, Hope, Ark. 71801.
(Originally published November 1986)

Photograph: Robert Lewis

A legitimate Golden Oldie, now well into its fourth decade, the Klipschorn, along with its designer, Paul Klipsch, are true legends in the field of high-fidelity sound. Nothing I could write would do complete justice to its description, so let me quote the words of a dear friend, long past, Howard Tremaine, who many years ago described the Klipschorn in his *Audio Cyclopedia* (Howard W. Sams & Co., 1959) as follows: "The enclosure is a low-frequency horn so folded that it may be placed in a room corner to utilize reflections from the floor and walls to improve the impedance match at the mouth of the horn and thus increase the response at low frequencies." Continuing Tremaine's description: "One advantage of using a horn at the low frequencies, compared to the use of a direct radiator mounted in a flat baffle, is that the horn efficiency is 10 to 50 times greater; and because of the acoustic loading, a given acoustic power may be generated with considerably less excursion of the loudspeaker diaphragm, thus reducing harmonic and intermodulation distortion."

The low-frequency horn is substantially exponential in its expansion rate and thus would have an acoustic path length of about 8 feet if unfolded. This horn is intended to cover the lower four octaves of the audible range, from about 32 Hz to 400 Hz, where the folded horn is crossed over to a midrange "squawker" that carries the range up to 6 kHz, where the response passes to a tweeter. Midrange and tweeter are also horn loudspeakers, and both are mounted behind a grille assembly placed on top of the folded bass horn.

The system is heavy. Oh, my, it is heavy. And it is big. But there is a reason for its size, and the reason is acoustic efficiency. Ten watts of music into this speaker will part your hair if you stand too close. One is soon reminded of the old horsepower adage: There's no substitute for cubic inches.

Because the Klipschorn uses the corner and floor of the room as a part of the bass reproduction process, the speakers must be placed in the corners for best reproduction. If the listening room does not have available corners or is very small, then, in my opinion, purchase of a Klipschorn system should not be contemplated. It simply needs room to sing.

Because of its bulk, the Klipschorn comes in two pieces, the bass horn and the midrange/tweeter unit. The instructions are clear, and no difficulty should be experienced in assembly or hookup. There are no controls or switches, and electrical connection is made to two well-marked terminals on the rear of the enclosure.

Because of the extreme efficiency of these speakers, you will not need a large power amplifier; 100 watts is more than adequate, and most listening will probably be done at levels below 5 watts. But the amplifier must be of high quality and have low noise. If it should have any hiss or hum, you will hear it with these speakers.

Measurements

The measured magnitude of impedance which the Klipschorn presents to an amplifier is shown in Fig. 1. The lowest value within the audio range is 4.5 ohms and occurs at 55 Hz, while the highest value is almost 10 times that amount, 42.3 ohms, at 2,155 Hz. The peak lies above the limits of the plot of Fig. 1, which is scaled to show the details of impedance throughout most of the audio range.

From the standpoint of amplifier drive requirements, a worst-case frequency appears to be around 5,200 Hz, where the phase angle lags by 70°, although the magnitude of impedance is 11 ohms. This can be seen in the complex impedance plot of Fig. 2. The many pig-tails in this plot are due to local impedance resonances. With the exception of the major bass resonance at 37 Hz, the majority of pig-tails are probably caused by acoustic reflections which occur in the bass, midrange, and tweeter horns. Figure 3 shows the complex impedance rescaled to show the midrange impedance peak at 2,155 Hz. This peak is not a smooth loop, but itself has several minor deviations in its peak range.

Fortunately, from the standpoint of amplifier drive requirements, none of these deviations can cause any problems whatsoever, as long as the power amplifier can drive 4 ohms at modest power. Because of this, and the fact that up to 25 average watts there is absolutely no change of admittance with drive level, I chose to omit the admittance plot for the Klipschorn. In this case, we do not need it.

Doing a complete set of acoustic performance measurements on the Klipschorn is a tour de force for any reviewer. This is a corner horn loudspeaker system; hence it requires a corner. How does one make free-field response measurements when there are corners? Paul Klipsch solved the problem by building an anechoic chamber with insertable corners. The Klipschorn is also intended to be listened to at ranges greater than 3 meters. The substantial size of this speaker and its geometry require that measurements be performed at such distance, and I have chosen 3½ meters, since this is the distance at which I listened to these units. Even assuming I could lift it, I could not haul the Klipschorn out of doors for lower frequency measurements since it needs corners to reproduce low notes. I puzzled over this problem for quite some time (more time than my incredibly patient editor should ever be forced to wait for a review) and then decided to resort to computer software and physics.

Figure 4 shows the measured free-field amplitude of sound pressure as a function of frequency for a constant drive voltage corresponding to 1 average watt into 4 ohms. The plot is corrected for an equivalent distance of 1 meter on axis relative to the front of the enclosure, although the actual measuring distance is 3½ meters.

Figure 5, the free-field phase response, is plotted in two sections. The midrange phase plot is corrected for a time delay of 11,980 µS, and the tweeter phase plot is corrected for a time delay of 10,308 µS The 1.672-mS time difference is caused by the physical offset between the tweeter and midrange.

The free-field sound is reasonably uniform from a lower cutoff of around 38 Hz to an upper cutoff of around 18 kHz. The irregularities both above and below the acoustic crossover at 6 kHz are caused by internal acoustic reflections from the drivers, horns, and grille assembly. The system is incredibly sensitive, producing well over 98 dB per watt at 1 meter. It is easy to see why the suggested amplifier rating is only 20 watts per channel. This system really will give the rated 104 dB SPL at a distance of 4 feet into a room. One watt into a Klipschorn will produce the sound level that 30 watts produces with many smaller loudspeakers. If one were to use the full 100 watts of drive for which the Klipsch is rated, the sound level would soar to migraine limits. Dropping a stylus on a record might break a lease, as well as some crockery.

The low-frequency response has some interesting surprises. A measured low-frequency roll-off below 38 Hz does not seem impressive; there are many smaller enclosures which measure as well. But something happens when this low frequency comes as a large-area wavefront whose boundaries are the walls of the room, rather than as a wavefront expanding spherically from a position in front of a wall. For one thing, the first impression one has is that the low end is deficient, because the low-frequency rumbling and grumbling of most systems, which many people associate with low-end reproduction, just isn't there. However, as one begins to really listen to the music and sound, one

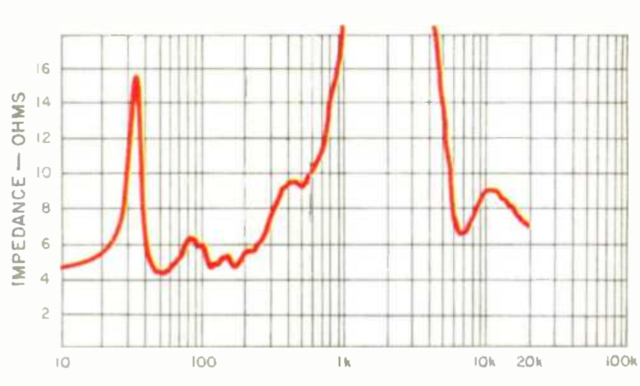


Fig. 1—Magnitude of impedance.

The Klipschorn is heavy! And big! But there is a reason—acoustic efficiency. Just 10 watts will part your hair at close range.

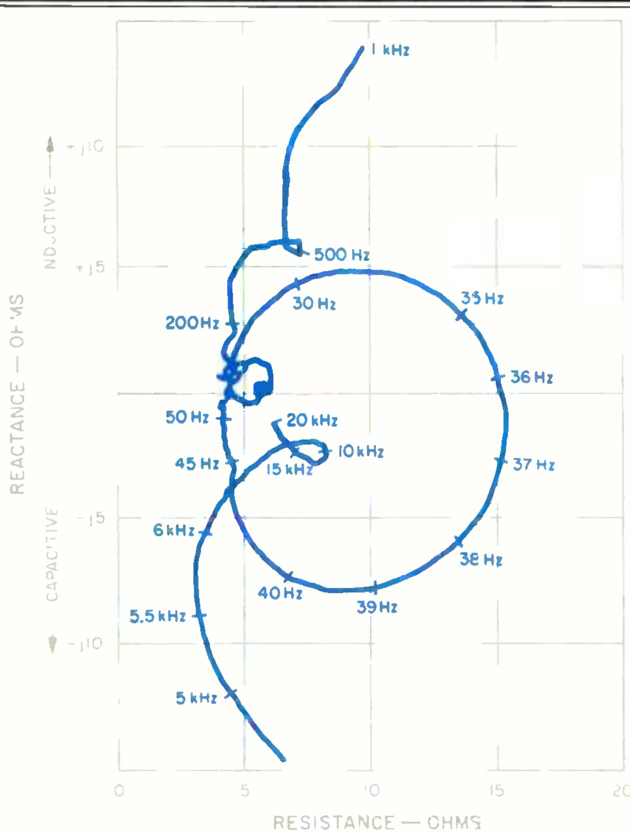


Fig. 2—Complex impedance; see also Fig. 3 for the range from 1 to 5 kHz.

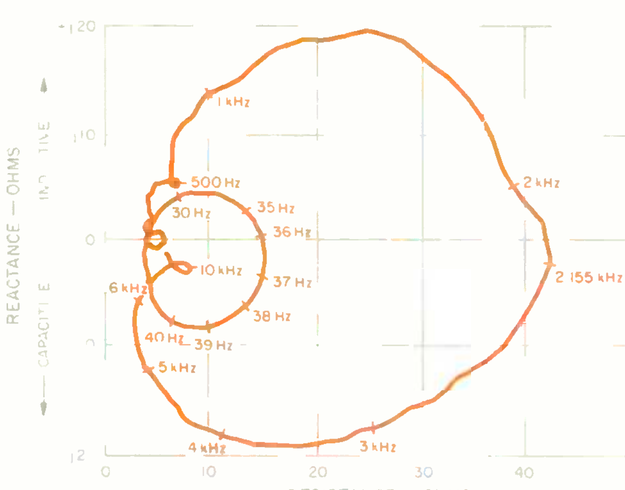


Fig. 3—Complex impedance. Note the rescaling from Fig. 2.

realizes that the deep bass is actually there and that it sounds natura', not overemphasized. Real low-end ambience in a room doesn't rumble; it's simply there as a pervasive ambience. So with the Klipschorn. To be sure, the horn unloads very rapidly as one progresses below 35 Hz and the woofer can be made to rattle with excessive drive at 10 Hz with no great amount of low-frequency musical content in the room, but above 35 Hz it is darn good.

The free-field frequency response measurement of Figs. 4 and 5 required some computer processing. Figure 6 is the 20 Hz to 20 kHz energy-time curve (ETC) of the Klipschorn, measured in the listening room at a distance of 3½ meters. The floor, ceiling, walls, and furniture reflections are present in this measurement. Figure 7 is a processed ETC in which everything has been removed except the direct sound from the Klipschorn. The frequency response measurements of Figs. 4 and 5 correspond to the ETC of Fig. 7. The expanded ETC of the free-field response of Fig. 7 is shown as Fig. 8. This departs from our conventional review format, which puts this expanded ETC at the end of the review.

It is instructive to compare the ETCs of Figs. 6 and 7 since they explain some of the audible midrange problems with this system. Consider Fig. 7. The first response, at 10.3 mS, is the sound from the tweeter, which is mounted up front on the grille. The second response, at 12 mS, is the sound from the midrange horn, whose compression driver lies back near the rear of the enclosure. Believe it or not, the small broad peak at 17.7 mS is the sound from the bass driver, which carries the frequencies below 300 Hz. Now consider Fig. 6. This is the complete sound, room and all. The multiple peaks at 12.5 and 13 mS are due to the tweeter sound which reflects off the side walls and the ceiling. At the listening location, we first hear the tweeter, then the midrange, then a staccato hit from the tweeter reflecting off the floor and ceiling, then a weak tweeter reflection from the side wall, and then the midrange reflecting off the floor and ceiling, with the rest of the room furnishings coming in several milliseconds later. Fortunately, the left and right channels are symmetric in this sound, since it is caused by the geometry of walls, ceiling and floor. In my earlier listening test, I felt that there was a problem with upper-midrange instrumental clarity, and I believe it is due to this effect. I infer from these actual room ETCs that the Klipschorn will sound best in a very large room with a high ceiling and a heavily carpeted floor.

There is another obvious item related to geometry, but due this time to the geometry of mounting the tweeter and midrange horns on the front of the enclosure. The ETC of Fig. 8 illustrates this situation. In Fig. 8, I have corrected the time scale to correspond to a microphone position which is 1 meter in front of the grille. The tweeter sound arrives at 3.7 mS but has an internal reverberation whose period is about 167 μS with a decay rate of about 9 dB per period. This causes the irregularities in free-field sound around 6 kHz, which is evident in Fig. 4. Since this frequency coincides with the acoustic crossover range, the reverberation may possibly be associated with the crossover process. The midrange sound first appears at 5.38 mS and shows a mild reverberation characteristic which pulls the energy out for a half-millisecond or so before it drops. Subsequent enclo-

Measured "free-field" using computer software, the K-horn is reasonably uniform from around 38 Hz to upper cutoff at 18 kHz.

sure reflections occur after about 6 mS. The first sound from the woofer is not on this measurement since it arrives about 8.4 mS after the sound from the tweeter.

What does it mean? A loss of clarity for those sounds which contain significant energy around 6 kHz, such as higher register female vocals and piano, but clean transient sound for both mid-register and extreme upper-register instruments such as some horns and triangle.

The 3-meter room response (which, for this speaker only, was actually measured at 3½ meters) is shown in Fig. 9. I measured only the on-axis response, not only since this is the recommended listening position, but also because a 30° off-axis response does not make much sense for a corner horn. This is the frequency spectrum of the first 13 mS of sound at the listening position and includes all the signals shown in the ETC of Fig. 6. The effect of early reflections is clearly evident in the plot of Fig. 9. Of particular interest is the fall-off of energy below 350 Hz. The reason is simple: The woofer sound doesn't begin to arrive until more than 8 mS after the sound from the tweeter, and is rejected by the time gate of the TDS measuring instrument. If I were to tune to the sound from the woofer, then the midrange and tweeter would be substantially reduced. In general, the 3-meter response is quite similar in character to the free-field response (which was taken at precisely the same physical location). The timbral balance of high, mid, and low portions of the spectrum is quite good. Only the time of arrival of those sounds will detract from sonic accuracy.

For reasons that may be obvious, I was not able to make a far-field turntable measurement of the horizontal and vertical polar energy response. Even if I had the services of King Kong to move the speaker, I would still need to rotate the whole room, walls and all. However, I was able to verify, by selected close-up microphone measurements, that the horizontal and vertical polar energy response was essentially smooth within ±15° of the normal listening position. This agreed with my earlier listening impressions; I was able to walk around the room over a significant range, without change in the level or tonal balance of the sound.

Measured harmonic distortion for the frequencies of 41.2 Hz, 110 Hz and 262 Hz is shown in Fig. 10. These frequencies correspond to the musical tones of E_1, A_2, and middle C, respectively. I chose middle C rather than A_4 (440 Hz) because the actual acoustic crossover from woofer to midrange is slightly above 250 Hz, and I wanted to measure the distortion for the same driver at all frequencies. Low bass (E_1) harmonic distortion progresses smoothly from a few tenths of a percent at 100 mW drive upward to near 10% at 60 average watts, with second harmonic slightly above third harmonic throughout the whole range. Mid-bass harmonic distortion at 110 Hz does not look at all like the low-bass distortion. Mid-bass harmonic level stays essentially uniform and of low level throughout the entire drive range, as the woofer really likes the acoustic load in this important frequency range and pumps out acoustic power with little distortion. Up near the crossover at middle C, the harmonic distortion again rises uniformly with drive level, although its total level is very low even at a thundering 100+ dB SPL at normal listening distance. If you like it loud and you like it clean, this is the speaker.

Intermodulation of middle C by E_1 (41.2 Hz), when both are mixed in equal proportions, is shown in Fig. 11. The magnitude of IM is impressively low when we look at the sound pressure levels which are involved. Music played at a 10-watt average level into the Klipschorn is reproduced at sound pressure levels which many other fine loudspeaker systems simply cannot reproduce, yet the IM remains below 4%. The nature of this IM is principally amplitude modulation of middle C by the lower tone up to about 50 average watts. At 100 average watts (ear-protection levels), the IM mea-

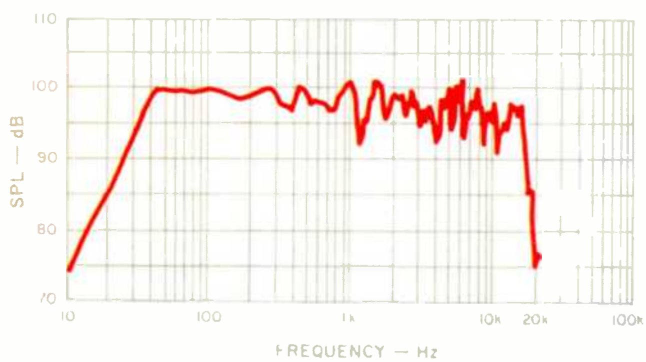


Fig. 4—Free-field sound pressure level for a constant voltage drive corresponding to 1 average watt into 4 ohms.

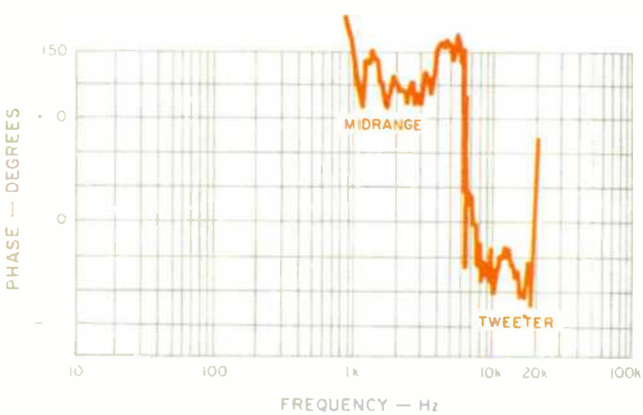


Fig. 5—Free-field phase response at 3½ meters. The midrange response is corrected for 11,980-μS time delay, while the tweeter response is corrected for 10,308-μS time delay.

Because the listening room is part of the low-bass reproducing system, the K-horn requires at least four feet of side walls in the listening room.

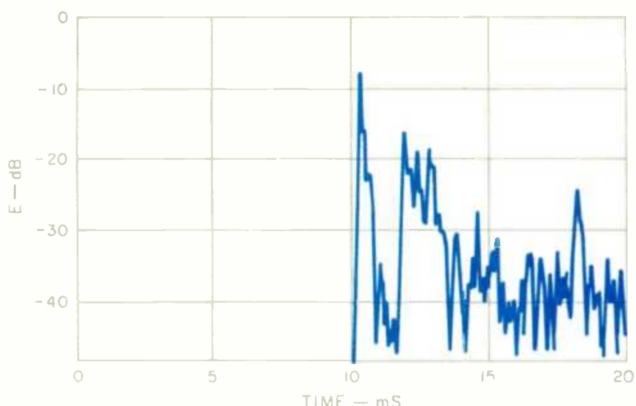


Fig. 6—ETC of the Klipschorn including early reflections from the room.

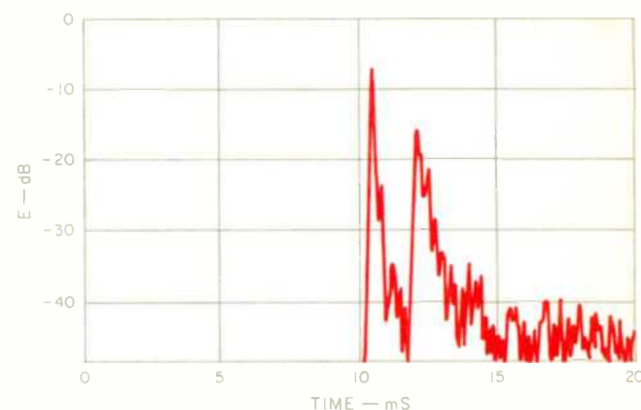


Fig. 7—ETC of Fig. 6 minus the room reflections.

Fig. 8—ETC of Fig. 7 with an expanded time scale and corrected for a 1-meter measurement position.

sures 12.88% and has picked up a phase modulation of 6° peak-to-peak on middle C, in addition to about 8% peak-to-peak amplitude modulation.

The result of the crescendo test is also impressively good. In this test, the ratio of sound pressure level to drive power is measured for selected musical tones. Perfection occurs when the SPL precisely tracks the drive power. In the case of the Klipsch, the tone of E_1 (41.2 Hz) slowly drops in relative SPL with drive power such that a 63-watt test level is 0.5 dB below the level which represents perfection, relative to a starting power level of 100 mW. The tone of A_2 (110 Hz) drops in a similar manner by 1.2 dB from a starting reference of 100 mW to a 63-watt maximum test level, while middle C slowly drops by 0.7 dB over the same range. The net effect will be an extremely mild softening of instrumental timbre as the sound pressure rises to high levels. This measurement also indicates that no discernible lateral shift of stereo image should occur with changes in drive level.

The same exemplary performance is maintained when a second musical tone is added to an existing tone. This means that stereo imaging should remain steady, with no instrumental wander caused by changes in musical dynamics throughout the useful intensity range of the Klipsch reproducer. In short, the Klipsch stays together regardless of what happens in the music.

Use and Listening Tests

A stereo Klipschorn reproducing system requires a room with two good corners. Period! If your listening room does not have left-channel and right-channel walls which come out in an uninterrupted fashion for at least 1½ meters from their respective corners, then forget it.

I am fortunate in that I have two such corners in my listening room. The geometry of the room also reasonably matches Klipsch's recommendations of a ratio of 1.00 to 0.618 for distance between speakers to distance from front wall to rear wall. The reason for all this fussiness becomes evident when you begin to listen to the system: The listening room is part of the low-bass reproducing system.

As mentioned above, the first impression one has is that the Klipschorn is deficient in low bass. This impression is visually reinforced by the massive presence of the system itself: "Anything that big should go right on downstairs in bass." But as one settles down for listening, it becomes apparent that the low end really is there. Not obtrusive, not rumbly, but there.

Many years ago, when listening to a similar pair of Klipschorns, I decided to find out how accurate the low end was. So I placed two high-quality condenser microphones outside my house, in a location where I could listen to the sound they picked up while viewing the same microphone location through a picture window that stretched between the two Klipschorns. It was only a matter of walking outside

Overall, the Klipschorn demands a great deal of respect as an accurate reproducer, surviving modern recording and electronic technology well.

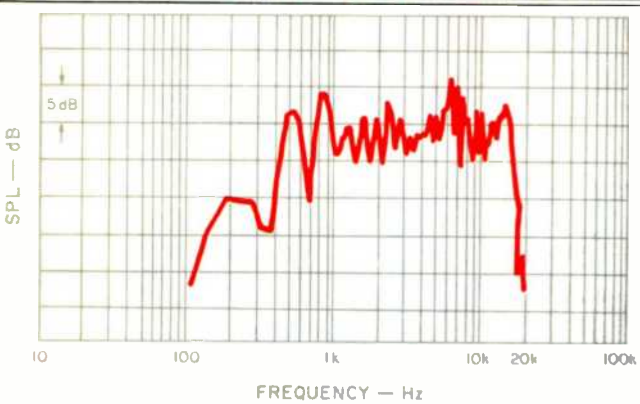


Fig. 9—Three-meter room response; see text.

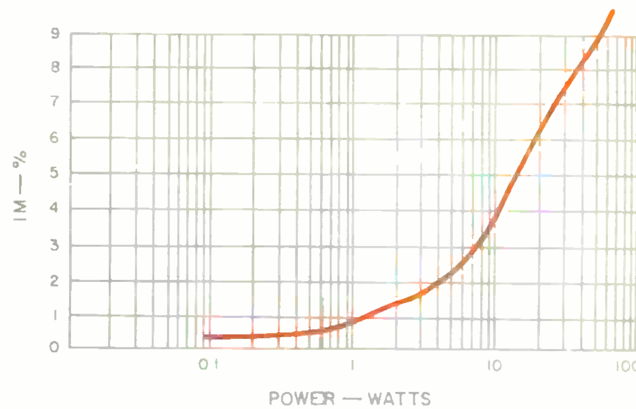


Fig. 11—IM of middle C (262 Hz) caused by mixing with E_1 (41.2 Hz) at equal levels.

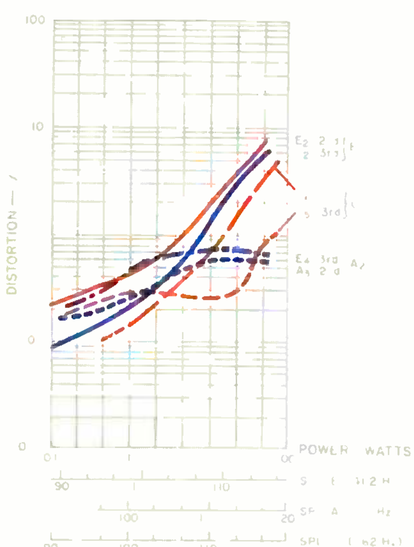


Fig. 10—Harmonic distortion for the test tones of E_1 or 41.2 Hz, A_2 or 110 Hz, and C_4 (middle C) or 262 Hz.

and listening, then walking inside and listening to compare the reproduced sound with reality. I could also switch between the K-horns and a pair of excellent speakers whose bass could shake the house on pipe organ; they made the K-horns sound thin by comparison. Then a funny thing happened. The sound of a slammed car door sounded like a slammed car door on the K-horns, but sounded like muffled "whumps" on the "wider range" system. The same with helicopter fly-overs (quite frequent where I used to live) and with the sound of distant traffic. I never forgot that experiment nor its ear-opening ramifications with regard to sonic accuracy versus measurement. Quite true, I have listened to many excellent subwoofers that could shake the walls at 10 Hz, while the K-horn produced little sound pressure even an octave above that frequency. But in my personal opinion, accurate percussive bass is a specialty which a properly set-up corner horn seems to have to itself.

Orchestral balance is also quite accurate; horns and strings stay put and are accurately placed on the stereo stage. Brass is brilliant and accurate on this system, and these instruments are so well placed that I felt I could point directly at each instrument. On the down side, to my ears, vocals, particularly female ones, seemed strident, and I could not get an accurate sonic illusion of piano, which always seemed larger than life, even at lower sound levels.

The usable listening area extends over much of the listening room, and one can move about freely without losing stereo balance as long as the speakers are at least 3 meters from your listening location. It takes a pretty good-sized room to get a good sound from the Klipsch system; a small room will probably produce sonic disappointment. This is not a speaker system you haul to a dormitory.

The Klipsch system has two additional sonic characteristics which warrant discussion. First, it is one of the few sound-reproducing systems which sound natural when one walks into an adjacent room. This is an interesting subjective illusion, one which I cannot explain. However, we have all had the experience of hearing a live musical instrument being played in an adjacent room; it still sounds natural and we can readily tell that it is not artificially reproduced. The piano recordings with which I had had trouble while in the listening room actually sounded "live in the next room" when I was in a room adjacent to the listening area. While others may disagree, that is the illusion I experience.

The second characteristic is the maintenance of timbral balance even when the sound is reproduced at substantially lower sound levels than would be normal for a given piece of material. Again, this is my personal opinion, and others may disagree.

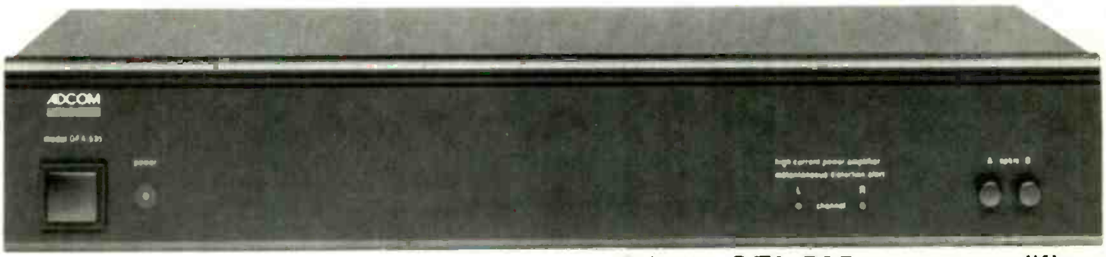
Overall, the Klipschorn is a Golden Oldie that survives modern recording and electronic technology very well. A bit jagged in the midrange, it still demands a great deal of respect as an accurate reproducer. *Richard C. Heyser*

"In its price category, the Adcom GFA-535 is not only an excellent choice; it's the only choice."

Sam Tellig, The Audio Cheapskate

stereophile

Vol. 10 No. November 1987



Adcom GFA-535 power amplifier.

The complete report:

Sometimes products are too cheap for their own good, and people don't take them seriously: the Superphon Revelation Basic Dual Mono preamp, Rega RB300 arm, AR ES-1 turntable, Shure V15-V MR cartridge, and the B&K ST-140 power amp. They can't be any good because they cost so little, right?

Wrong, of course.

Adcom appears to be having the same problem with their $299.95 GFA-535 amp. Credibility.

Now if this amplifier were imported from England and sold for $599.95, then maybe it would be taken seriously. And highly praised, no doubt.

For the baby Adcom is one of the finest solid-state amps I have heard. No, not the best; I'm not sure what *is* the best. But it's an amplifier that is so good for so little money as to be practically a gift.

Actually, when Rob Ain from Adcom called, I was about as enthusiastic about the GFA-535 as you were before you finish reading this piece. But Rob insisted, "You've gotta hear this amp."

He brought it over the next day, along with the GFP-555 preamp ($499.95), and we put both pieces into the rest of the system: a Shure Ultra 500 in a Rega RB300 arm on an AR ES-1 table, with Quad ESL-63 speakers on Arcici stands. Then we chatted for a half hour or so while the electronics warmed up.

And then, simultaneously, the two of us decided to shut up and listen.

"I've never heard the Quad ESL-63 sound better," Rob said. Of course, he was hardly an impartial observer, but the sound was extraordinarily clean, detailed, and musical. If it wasn't the best sound *I* have ever heard from Quads, it was pretty close.

This humble $300 amplifier was driving a pair of very revealing $3000 speakers and giving a very good account of itself. (We listened first to some Goran Sollscher classical guitar.)

"So how come this product isn't flying off the dealers' shelves?" I asked Rob.

"I don't know. Everyone wants the GFA-555 with 200 watts per channel. Including people who don't need it."

"Does the GFA-555 sound any better?" I asked.

"It's our aim to have all our amps sound pretty much the same. You pay more money, you get more power."

Rob pointed out that while the GFA-535 is rated at 60Wpc, it puts out more like 80. And while I did not do any measurements, my experience with other amps tells me Rob's right. I suppose Adcom doesn't want to steal sales from its GFA-545, rated at 100Wpc and selling for $200 more.

After a couple of hours, Rob left, grinning from ear to ear, and I later sat down to listen alone. True, when I tried certain Telarcs and pushed hard I could get the amplifier to clip—two LEDs quickly light up (very useful). But the Quads were running out of the ability to use the power anyway. My first impressions

2

VELODYNE ULD-15 POWERED SUBWOOFER

Manufacturer's Specifications

System Type: Sealed-box woofer with separate feedback-controlled power amp and electronic crossover.

Driver: 15-in. (38-cm) woofer with accelerometer and preamp.

Frequency Response: 20 Hz to selected crossover frequency, ±3 dB.

System Distortion: Less than 1.0% at half power, 20 to 85 Hz.

Amplifier Power: 350 watts.

Input Impedance: 20 kilohms.

Input Sensitivity: Variable, 300 mV or more for maximum output.

Crossover: Frequencies independently selectable for high pass (36 to 212 Hz) and low pass (58 to 193 Hz) via solder-in resistors installed by dealer. High-pass slope (6 or 12 dB/octave) set by dealer. Shipped set for 85-Hz, 12-dB/octave operation.

Dimensions: Speaker, 18¼ in. H × 22½ in. W × 17 in. D (46.4 cm × 57.2 cm × 43.2 cm); electronics, 3½ in. H × 17 in. W × 11 in. D (8.9 cm × 43.2 cm × 27.9 cm).

Weight: Speaker, 57 lbs. (25.9 kg); electronics, 19 lbs. (8.6 kg).

Price: $1,795.

Company Address: 1746 Junction Ave., San Jose, Cal. 95112.

(Originally published November 1987)

Photograph: ©1987, Bill Kouirinis

The Velodyne ULD-15 is a complete mono subwoofer system that will extend the response of stereo speakers to below 20 Hz with an unprecedented low level of distortion. The down-firing 15-inch woofer is housed in an attractive and amazingly small cabinet. The separate crossover/amplifier is housed in a rack-mount style chassis 3½ inches high. If you like clean, low bass from your audio system, this is a painless way to get it.

In addition to the 15-inch model tested, Velodyne makes three other servo or feedback woofers, an 18-inch model for $650 more than the ULD-15 and two 12-inch models, for $600 or $900 less. They differ mainly in the sophistication of their electronics and in their output capability. A single ULD-15 is a good choice because it is a match for anything found in live music short of cannon fire. All four models, however, boast distortion figures closer to those of amplifiers than those usually associated with speakers. Velodyne points out that woofer distortion is highly audible because, at low frequencies, hearing sensitivity is greater for the harmonics than for the fundamental.

Wide-bandwidth negative feedback is the key to the Velodyne's performance. Despite its name, negative feedback is very positive in its results. In non-electroacoustic form, it is very familiar: When we perform any physical activity, such as walking, we rely on our senses to constantly correct and refine our motions. This corrective action is a form of negative feedback. A negative feedback system consists of a power source that is controlled by both an input and a sensor from the output. Our body is the power source, the desire to walk down a path is our input, and sight is the primary sensor to keep us on the path.

An ordinary woofer receives a signal from the power amplifier and "woofs" only as accurately as its suspension and the voice-coil drive force allow. A negative-feedback woofer, however, senses its own output and uses this information to keep itself on the path dictated by the input. Negative feedback is a continuous process, correcting for every nuance of the input waveform.

The power amp is a woofer's power source, so it must be made part of any feedback system. In other words, negative feedback from the output sensor must be applied at the input of the power amp, thus enclosing it in the loop along with the woofer. The result is an amplifier/woofer system that can reduce distortion of an "open-loop" system by a factor of 10 or more.

Figure 1 is a block diagram of the Velodyne system, showing the active crossover filters, a limiter, and the components of the feedback system. The limiter is a gain-reduction device which operates only when the input signal is great enough to cause amplifier clipping or excessive cone excursion. In this feedback system, allowing either condition to occur could cause more than the usual amount of distortion. While it's clipping, the amplifier would be unable to respond to feedback and would lose track of the cone's position; due to feedback, excessive excursion would make the amplifier clip.

The purpose of the box labelled "Compensation" is to prevent the negative feedback from becoming positive at any frequency. With a phase shift of 180°, the feedback signal would begin adding to the input, quickly producing a full-power oscillation. This happens most readily at frequencies in the range above 1 kHz, where the coupling between woofer and sensor is affected by resonance and propagation delays. To prevent this, the compensation network gently attenuates high frequencies and, with them, high-frequency feedback. When the loop finally gets 180° out of phase, the signal is too far attenuated to produce positive feedback. This may seem complicated, but it's a technique that has been used in amplifiers since the 1930s.

The obvious way to sense the woofer's output would be to use a microphone. Unfortunately, the time it takes for the signal to propagate only a few inches through air would

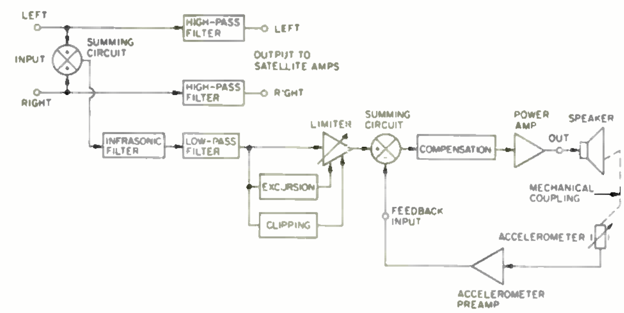


Fig. 1—Block diagram of the ULD-15 (see text).

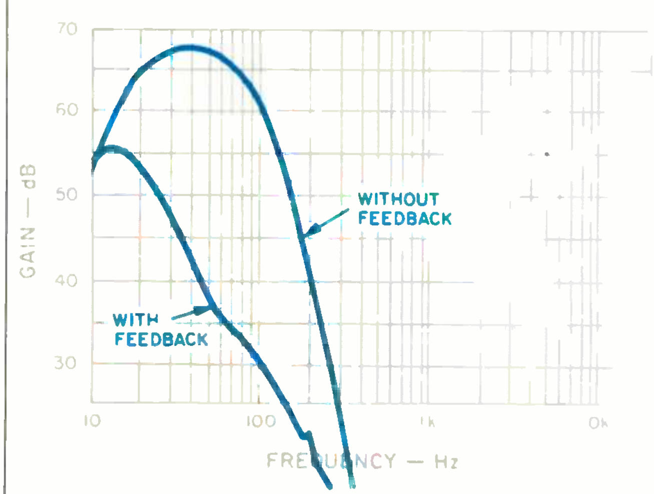


Fig. 2—Amplifier frequency response, measured from left input to speaker terminals, with feedback loop both connected and opened.

The difference between the curves shows the amount of feedback available for reducing distortion.

cause enough phase shift to require excessive compensation, resulting in reduced high-frequency feedback. The method chosen by Velodyne is to mount a piezoelectric accelerometer to the voice-coil form. This lightweight crystal pickup generates an electric charge proportional to its acceleration. An amplifier located on the speaker frame converts the tiny charge variation to a voltage which is fed back to the power amp's input.

Acceleration-sensitive pickup has one advantage. A flat frequency response for a small radiator (even a 15-inch cone is small in relation to the 11-foot wavelength of a 100-Hz tone) results from equal peak *acceleration* at all frequencies. You might think the cone would have to move the same *distance* at all frequencies for a flat response. Not so. It must increase its stroke to maintain output at low frequencies. If an acceleration pickup were not used, an additional conversion stage would have to be inserted in the feedback path to make the pickup acceleration-sensitive.

In a small box, a 15-inch woofer would normally have a peak around 50 Hz. Negative feedback gives the ULD-15 flat response.

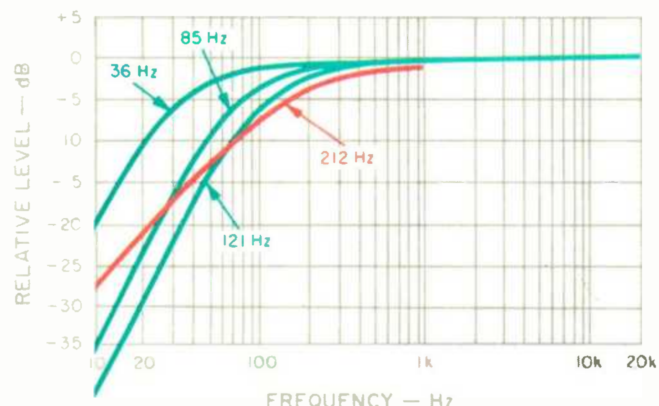


Fig. 3—Some high-pass crossover response curves available by changing resistors (see text). Note that the 212-Hz curve uses the optional 6-dB/octave slope; the other curves use the 12-dB slope.

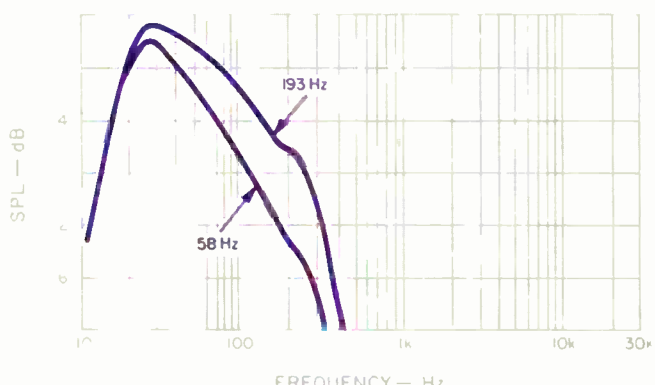


Fig. 4—Acoustic output for 58- and 193-Hz low-pass settings with 0.01 V input, measured at 1 meter with speaker placed on floor.

Attention to details is crucial in the design and manufacture of a complex system—this is where the Velodyne engineers have done excellent homework. The pickup is located on the edge of the voice-coil (on which Velodyne has a patent); a proprietary process determines the optimal location on the voice-coil's edge for every driver. In an additional, custom hand-tuning process, small weights are applied to the cone and patterns of small holes are punched into it. This stabilizes the servo system by controlling the alignment of breakup modes. (The holes are later covered by a dome, so they do not cause air leaks.) What results from this tweaking process (on which a patent is pending) is a rock-stable system with feedback effective out to at least several hundred hertz.

Since negative feedback makes the system behave more like the sensor (very flat and with very low distortion) than like the speaker/enclosure itself, a useful size trade-off can be made. In Velodyne's small box, a 15-inch woofer would normally have a peak around 50 Hz and poor low-frequency extension. With negative feedback the response becomes flat, but the trade-off is that more amplifier power is required. Even in the Velodyne's 2½-cubic-foot box, the driver can handle only a certain amount of power at extremely low frequencies before its excursion limits are reached. As long as this power is available, the small box is not a practical liability. The Velodyne amplifier is indeed powerful enough to push the driver to both its low-frequency excursion limits and its high-frequency thermal limits.

The electronics package contains all the electronic circuitry except the accelerometer preamplifier. It is a basic but attractive box made of brushed, black-anodized aluminum, with heat-radiator fins forming the sides. The top is fastened with machine screws that go into threaded inserts. Though always a sign of class, the machine screws are especially helpful here, because the top must be removed if the crossover frequency or slope needs changing.

The well-crafted and roomy interior houses three circuit boards, a large toroidal transformer, a single power-supply filter capacitor, and a few small components. The use of a single power supply, rather than the conventional plus and minus supplies, is made possible by using two amps in bridged mode. There is no output coupling capacitor, yet the output terminals are at ground potential when there is no signal. The single power-supply capacitor stores an impressive 40 joules of energy.

The electronics package (which must be plugged into an a.c. outlet) is connected to the speaker box by user-supplied speaker wire and by a four-pin DIN extension cable for the feedback sensor. Since the primary left and right signals must also feed into this unit (and come out again, if its built-in crossover is used), it makes sense to place the unit near the other electronic components of the audio system. Fortunately, Velodyne supplies a 24-foot DIN extension cable, allowing considerable flexibility in placing the speaker box.

The electronics' front-panel level control can be switched to control the subwoofer only or the entire stereo signal. In audio systems with accessible preamp outputs and amp inputs, the Velodyne's level control can be used to match the subwoofer's output to that of the rest of the system. For use with receivers or integrated amps having no such jacks, the Velodyne's electronics can be connected to a tape monitor loop and its level control switched to "Master" to control overall volume; in this case, the satellite speakers' output would be adjusted to match the Velodyne's by using the receiver's volume control. As the Velodyne's electronics have no tape monitor loop, restoring the receiver's tape monitor capability would require an external switchbox.

The ULD-15 is shipped with its crossover set at 85 Hz, which will work fine in most applications. However, the low- and high-pass frequencies and high-pass slope can be adjusted independently. Changing the frequency requires changing four soldered-in resistors for the high-pass section and one resistor for the low-pass section; to change the high-pass slope to 6 dB/octave, one simply removes two resistors. All this is normally done for the customer by Velodyne dealers, but it would not be too hard to do for oneself. When the electronics' cover is removed, it reveals clearly designated, forked terminals for the resistors.

The ULD-15 bettered its power specification with ease, producing 392 watts from 20 to 100 Hz with about 0.3% distortion.

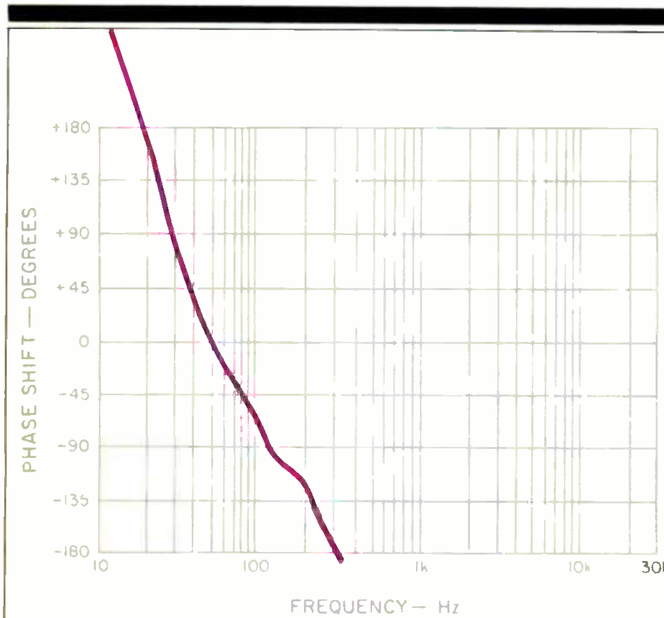


Fig. 5—Acoustic phase response for 193-Hz. crossover, measured at 1 meter with speaker placed on floor.

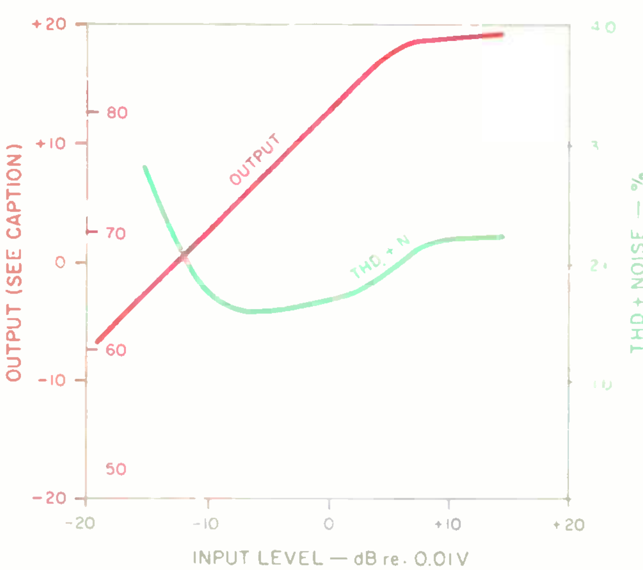


Fig. 6—Linearity, using 20-Hz signal, showing both output and THD + N vs. input voltage. Output curve shows both amplifier power (outer scale, in dB re: 1 watt) and acoustic output (inner scale, in dB SPL measured at 1 meter in half-space). Note flattening of both curves as limiter cuts in at high signal voltages. Apparent rise in THD + N at low levels is due to ambient noise (see text).

The sealed woofer box is finished on all sides in oiled walnut veneer. It is a sturdily built and rather attractive piece of furniture.

Measurements

The ULD-15 is an integrated electroacoustic system. It would normally be evaluated as a black box, from RCA-jack input to acoustic output. I decided to go a little further and poke around inside the loop. First I measured frequency response, with feedback, from the left input to the speaker terminals (Fig. 2). The speaker terminals are inside the loop, so this plot shows the amp output response that will produce the rated acoustic output range of 20 to 193 Hz. An increase in the amplifier output at low frequencies, to compensate for the small box, is evident.

Also plotted in Fig. 2 is response at the speaker terminals with the feedback cable unplugged. This curve results from the input signal passing through the crossover band-pass filter, the compensation filter, and the amplifier. (Refer back to Fig. 1, the block diagram, as necessary.) Note the difference between the two curves: it is a measure of the feedback available for reducing distortion and flattening response of the acoustic output. Note that from 30 to 200 Hz, there is at least 20 dB—and mostly more—of feedback. This translates to more than a 10-to-1 reduction in distortion over this range.

Let's look at what the two curves tell us about what is happening at 20 Hz. The 11 dB of feedback will correct the speaker's natural response roll-off quite closely, but what about the harmonics generated at 40, 60, 80, and 100 Hz? They fall in the range where an average of 30 dB of feedback is in effect; this results in a distortion reduction of 30 dB, or 32 to 1! This is not an exaggeration if the accelerometer is linear: If we start with 10% distortion in the driver, we end up with 0.32% for the system.

The power amp is designed specifically to drive the ULD-15 woofer/enclosure, and that would be its ideal test load. The trouble is that no woofer would be likely to survive continuous maximum-power testing. The woofer impedance was measured as 8.1 ohms at 20 Hz, rising to 45 ohms at the 48-Hz resonance and dropping to 7.7 ohms at 100 Hz. An 8-ohm load resistor was chosen as a substitute load for power testing. A minimum of 392 watts, with distortion at around 0.3%, was recorded over the range from 20 to 100 Hz. This betters the power spec handily. Remember, the distortion observed here is really of no concern—once the driver is reconnected, the amp will be inside the loop and subject to distortion reduction by the speaker feedback.

Conventional measurements showed an input impedance of 24 kilohms or more. The crossover's line outputs clipped at 7.5 V, with 0.2% THD + N. Distortion at the standard output level of 2.0 V was 0.064% maximum, from 20 Hz to 20 kHz. Gain through the high-pass section was unity unless the level control switch was set to "Master," in which case gain was variable to 21 dB.

Figure 3 plots some of the high-pass filter curves available. The 12-dB/octave slopes are reasonably accurate in frequency, but they sag more at cutoff than the 3 dB of the common Butterworth designs. This response shape is intended to help the subwoofer blend well with typical speak-

The limiter reduces gain only at frequencies and amplitudes where cone excursion would otherwise become excessive.

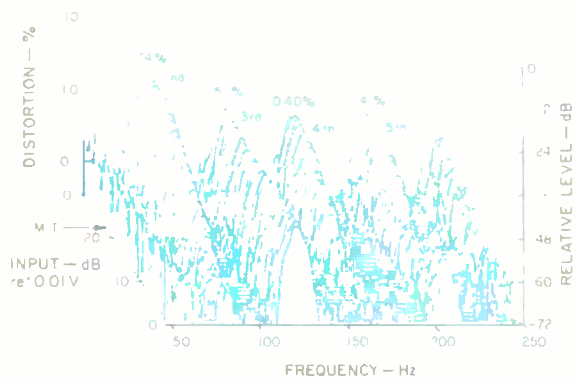


Fig. 7—Harmonic distortion for the musical tone E_1 (41.2 Hz).

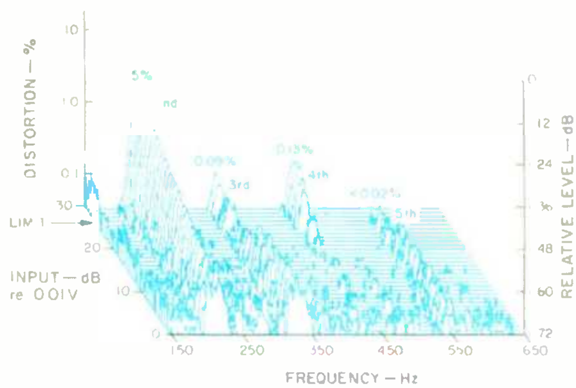


Fig. 8—Harmonic distortion for the musical tone A_2 (110 Hz).

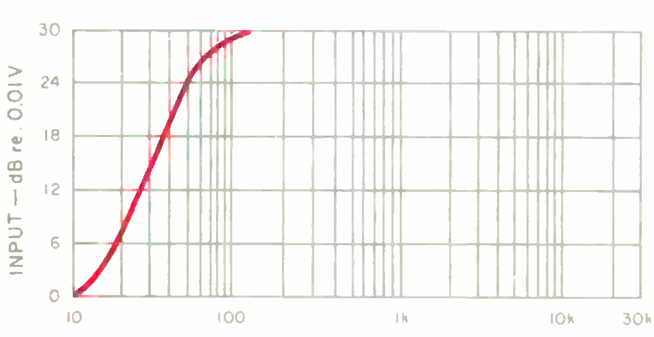


Fig. 9—Limiter action. Input voltage needed to cause a given level of compression varies with frequency to limit cone excursion (see text). Shown here: Voltage vs. frequency for 1-dB compression.

ers which have underdamped response at cutoff. For those speakers which are not underdamped, setting the high-pass section at a lower frequency might be just right.

My first acoustic output measurement was of frequency response (Fig. 4). For the test, I placed the ULD-15 on the floor, the condition for which the down-firing woofer is designed. This differs from my usual anechoic measurements, mainly in that it increases acoustic output by 6 dB for all frequencies in the ULD-15's range. Curves are shown for both the crossover's minimum (58-Hz) and maximum (193-Hz) low-pass filter settings. Their shapes are determined almost entirely by the response of the infrasonic and low-pass filters preceding the amplifier/woofer feedback system. The response of these filters, and of the high-pass filters in the crossover, is chosen by Velodyne to give the best blend between the ULD-15 and the upper-range speakers in use.

Phase response of the acoustic output is plotted in Fig. 5 for the 193-Hz low-pass setting. The phase shift is very much like what one would measure from a simple electronic circuit having the same frequency response as the ULD-15. This means the ULD-15 is a minimum-phase device and will interface predictably with its companion speakers.

Two plots are given in Fig. 6. The output curve (relating to the left-hand scales) shows how the amplifier and the acoustic outputs track increases in 20-Hz input only up to the point where the limiter kicks in. The limiter reduces gain to prevent excessive cone excursion at the frequencies and amplitudes where it predicts this problem will occur. (The limiter's prediction was right. I observed over ½ inch of peak-to-peak excursion at the point where it kicked in.) The THD + N curve is for the acoustic signal. Note that distortion holds at a constant 2.2% after the limiter is activated. The apparent increase in THD + N at very low levels is not an increase in THD but is really a measure of background noise in my lab; the mike was picking up ambient sound as well as true distortion products.

Figure 7 shows a "3-D" plot of the harmonics of a 41.2-Hz signal. This frequency is the lowest note of a normally tuned string bass. Even with feedback, the usual distortion components are all there—but notice their values: There is less than one-tenth as much distortion as one usually finds in speakers. Furthermore, the upper harmonics, which are at more easily audible frequencies than the lower ones, have the lowest amplitude of all. (The point at which the limiter begins to hold the output steady is indicated on the scale for input level.)

The "3-D" plot of the harmonics of a 110-Hz signal (Fig. 8) is just as remarkable. Only the second harmonic is worth noting. The absence of upper harmonics is a testimony to the wide bandwidth over which the ULD-15's feedback is effective.

If input drive is continuously increased, there will be a point, for any speaker, where the output cannot increase proportionally. This compression can take two forms. The first, waveform compression, squashes the shape of the sine wave and generates harmonics. In speakers, this is often a result of overexcursion. The other type is envelope compression, where the waveform remains sinusoidal (and thus does not sound distorted) but fails to increase in

The quality, if not the magnitude, of Velodyne's achievement is up there with Dolby noise reduction and the Compact Disc.

amplitude. Voice-coil heating and restricted vent airflow are common causes of envelope compression. Compression results in most speakers from both forms of distortion.

For the ULD-15, virtually all of the compression is of the envelope type and is produced by the predictive limiter. Figure 9 shows how this limiting action varies with frequency and input voltage for a given degree of power compression. The lowest frequencies require the greatest cone excursions, so they activate the limiter at a lower level. The audible effect of this compression is not distortion or bottoming, but rather a thinning out of the low bass as volume is raised to high levels. An important point is that Velodyne's limiter only allows use of the driver's full *clean* capability.

Use and Listening Tests

Blending a subwoofer with a satellite system can be thought of as either a problem or an opportunity, depending on one's point of view. The problem is that separating the signal can lead to a discontinuity of sound in the crossover range. The opportunity is that a separate subwoofer affords wider options for dealing with room acoustics. I see subwoofers from the opportunity side.

The satellites can be fairly small full-range speakers that allow placement for best imaging without regard to the best bass response. The subwoofer, on the other hand, can be placed wherever it yields the best bass, with almost no effect on imaging. The subwoofer level control effectively becomes a broad equalizer that helps compensate for room modes and losses through walls. Often, as is the case with the Velodyne, crossover points can be shifted to match the low-frequency capability of whatever speakers are used as satellites with the ULD-15. And because the low- and high-pass points are independently selectable, they can be overlapped or underlapped to deal with dips or peaks in the room's acoustic response at the crossover point. Also, flipping the subwoofer's polarity may improve results. The idea is not to try assembling a system that will be perfect in an anechoic chamber, but to tune the system to an imperfect room.

What kind of equipment is needed to find optimal subwoofer settings? Instruments can speed things along, but I always end up making final adjustments by ear using music. An excellent piece, recommended to me by Ivan Berger, is "You Look Good to Me" by the Oscar Peterson Trio on *We Get Requests* (Verve CD 810 047-2). I believe trial and error, guided by careful listening, is the best way to set up a satellite/subwoofer system.

First, one should find the best positions for the satellites, then place the subwoofer between them, if possible The crossover frequency should be set as low as possible, considering the bass capability of the satellites. Moving the subwoofer to different locations in the room will enhance and suppress different room modes—try for a balance. If a peak or dip in the crossover range is suspected, change the polarity, the frequency, or the overlap.

I followed this procedure in my listening room, using Magnepan MG-IIIa speakers as satellites. Interfacing a subwoofer with bipolar panel speakers such as these is considered by many to be very difficult or impossible. (Magnepan suggests using the MG-IVa, which has additional bass panels, if one requires more low-frequency output.) I ended up with a virtually seamless blend by placing the subwoofer between the satellites and against the wall behind them. The crossover was set for 85 Hz, high pass, to the Magnepans and 100 Hz, low pass, to the Velodyne. Polarity reversal was required.

The sound of the combined system retained all of the clarity and depth of image that properly set-up Magnepans are known for, and it added a previously unheard octave of low bass. Loudness capability, I would estimate, was increased by 4 dB. Gone was the feeling of strain I used to hear from my system at high signal levels. Although my 1,200-watt amplifiers were approaching clip level by the time I sensed this strain, I am sure its absence was due to the fact that the Velodyne's crossover had relieved my panel speakers of having to handle bass below 85 Hz. Deep bass from drums and organ pedals sounded precise and open rather than overly tight. Perhaps this subwoofer blends so well with satellites because there are no buzzing harmonics to call attention to it.

I can't say I've never heard a subwoofer that equals this one, but I've never heard a comparable one that measured less than 15 cubic feet. Still, I have minor quibbles or, more properly, observations about the ULD-15. On loud rock, the absence of distortion in this speaker may be responsible for reduced "punch" and "attack." Perhaps most rock is mixed on, and intended for, lesser speakers. Also, there was a bit of a "thud" tonality on many instruments, perhaps traceable to the rising low end of the ULD-15. Velodyne claims this is intentional and designed to compensate for roll-offs in the recording chain. I'm not sure I want my woofer to speculate on this matter, but if any excessive bass is heard, it can be removed by repositioning the woofer or by cutting the bass control a tiny bit. Asking for extremely low bass at high levels can activate the "keep it clean" limiter, thus altering spectral balance between subwoofer and satellites. If this is a problem, the solution is simple: Buy a second ULD-15.

The Velodyne subwoofer is one of those rare components I can recommend to almost anyone. A system based on the right two-way satellites and a ULD-15, plus a receiver and CD player, can cost as little as $2,100 and still knock your socks off if you set it up carefully. The next step is to use some of the more expensive "mini-monitor" speakers with the Velodyne. Although any expensive speaker's bass may be improved with the ULD-15 large panel speakers such as those from Magnepan or Quad generally benefit most.

The Velodyne subwoofer is the most interesting product I have reviewed to date. When asked about 20-Hz bass response and distortion, audiophiles traditionally mumble something about "no musical information below such and such" and change the subject. The craftsmen who built church organs in centuries past did not agree with this opinion, and they provided pipes with output down to 16 Hz. Velodyne recognized the problem of reproducing low bass and engineered a solution. This feat required multi-disciplinary ability, intuition, craftsmanship, organization of priorities, and common sense. I rank the quality, if not the magnitude, of this small California company's achievement up there with Dolby noise reduction and the Compact Disc.

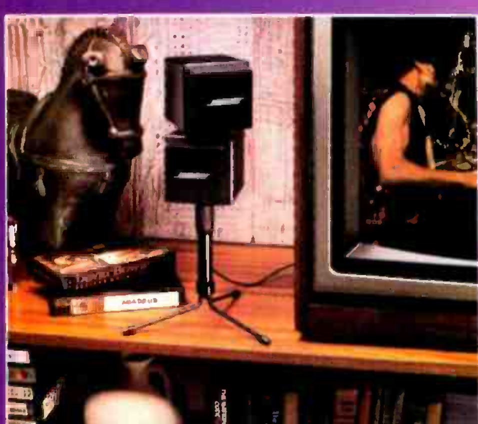
David L. Clark

“Superb sound and virtual invisibility.”

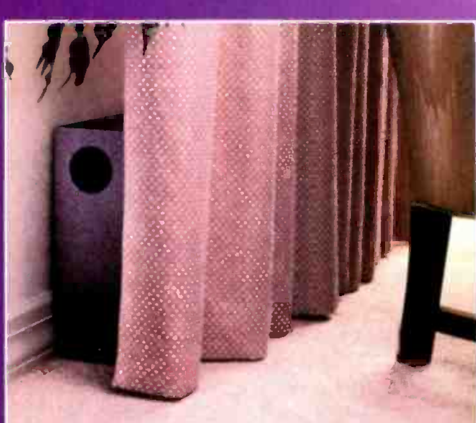
—*Stereo Review*, Julian Hirsch

Acoustimass® array with optional mounting accessory for unobtrusive placement.

The Bose® Acoustimass speaker system.



Both arrays are equipped with magnetic shielding for high-fidelity video listening.



The system's heart—the Acoustimass® module—can be completely hidden, providing *virtual invisibility.*

3

KEF 107 LOUDSPEAKER

Manufacturer's Specifications
System Type: Floor-standing, three-way, with coupled cavity woofer enclosure and external equalizer.
Drivers: Two 10-in. (250-mm) woofers, 5-in. (127-mm) midrange, and 1-in. (25-mm) dome tweeter.
Frequency Range: 20 Hz to 20 kHz, ±2 dB (equalized).
Sensitivity: 90 dB SPL at 1 meter for 2.83 V rms input.
Maximum Output: 112 dB SPL on program peaks under typical listening conditions.
Crossover Frequencies: Not specified.
Impedance: 4 ohms, resistive, 20 Hz to 20 kHz.
Recommended Amplifier Power: 50 to 300 watts into 4-ohm resistive load.
Dimensions: 45.9 in. H × 13 in. W × 17.6 in. D (116.5 cm × 33 cm × 44.8 cm).
Weight: 99 lbs. (45 kg).
Price: $4,800 per matched pair with KUBE equalizer.
Company Address: 14120-K Sullyfield Circle, Chantilly, Va. 22021.
(Originally published February 1988)

The KEF 107, the flagship model from KEF's "Reference Series" of loudspeakers, is primarily a high-end audiophile system. However, with its very high power-handling capacity, extended low-frequency response, and high maximum acoustic output capabilities, it will do justice to any type of program material from chamber music to hard rock. It represents the end result of many years of research and development and the pioneering use of computer-aided design and testing in the manufacture of loudspeakers and systems.

In early 1983, I had the pleasure of visiting the KEF plant in England, where I talked extensively with Laurie Fincham, KEF's technical director, and with KEF's founder and managing director, Raymond Cooke (whose position is equivalent to that of president of an American firm). I came back very impressed with the company, its philosophy, and its products. When given the opportunity to review one of KEF's high-end speaker systems, I jumped at the chance.

The KEF 107 is a three-way, modified direct-radiator system supplied with a line-level equalizer. The equalizer, called the KUBE (which stands for KEF User-variable Bass Equalizer), is used primarily to adjust the low-frequency characteristics of the system to the listening environment and also to equalize a moderate upper-midrange peak exhibited by the unequalized system. The equalizer is an

Fig. 1—Cutaway view of the KEF 107. Note the dual woofers, each firing from a sealed enclosure into a shared, ducted cavity, and the rod connecting the two woofer magnets. (See text.)

integral part of the system design and not just an after-the-fact addition to add some extra sales appeal.

The two low-frequency drivers are mounted in a modified version of the vented-box enclosure, called a "coupled cavity configuration" by KEF. This type of cabinet provides a band-pass type of frequency response because the acoustic output from the enclosure comes only from the vent.

The mid and high frequencies are radiated from a separate high-frequency enclosure or head assembly that is mounted on top of the low-frequency enclosure. This separate enclosure, which also contains the midrange/tweeter crossover, is shaped to minimize mid- and high-frequency diffraction effects; this ensures smooth on- and off-axis frequency response. The head, which is detachable, is connected to the bass cabinet with a gold-plated XLR connector. The head assembly can be rotated roughly ±30° to optimize coverage of the listening area. This method of handling the mid and high frequencies continues a successful approach first offered in 1976 by KEF in the Model 105.

KEF uses "conjugate load matching" in the crossover so that the system's input impedance, over the range of 20 Hz to 20 kHz, is essentially resistive, with a value of about 4 ohms. This load is very easy for any amplifier to drive. The lower value of impedance essentially doubles the voltage sensitivity of the system and raises the acoustic output, as compared to an 8-ohm system.

While KEF does not specify crossover frequencies, measurements revealed that the system is crossed over at about 160 Hz and 3 kHz and that the respective drivers' acoustic outputs are essentially in phase at crossover, thus minimizing lobing error.

The highs are radiated by a ferrofluid-cooled dome tweeter. The tweeter is a refined version of KEF's T33 high-frequency unit, which is also used in other KEF systems. The mid-frequencies are radiated by a polypropylene-cone midrange driver with an effective radiating diameter of about 4 inches (100 mm). This driver is an improved version of KEF's very fine B110 midrange unit, with higher sensitivity and power handling. The mid-frequency driver is used down to the relatively low frequency of 160 Hz, because the band-pass vented-box low-frequency system will not easily go any higher in frequency. The wide range covered by the midrange driver (160 Hz to 3 kHz, over four octaves!) opens up the possibility of intermodulation of the upper-midrange frequencies by high-amplitude signals in the 150- to 300-Hz range.

The low frequencies, below 160 Hz, are generated by a special form of vented box called a band-pass vented-box enclosure. This modified form of vented-box cabinet differs from the standard vented box because all the sound generated is radiated from the vent only. No sound comes directly from the drivers because they are contained within the sealed-box enclosure. This enclosure has a band-pass type of frequency response that, unassisted, covers only a response of roughly one to two octaves. KEF's technical director, Laurie Fincham, completely described this type of system at the May 1979 Audio Engineering Society Convention in Los Angeles (Preprint 1512, D-4, "A Bandpass Loudspeaker Enclosure"). Additional information can be found in a more recent paper by E. Geddes and D. Fawcett, "Bandpass Loudspeaker Enclosures," which was presented at the November 1986 AES Convention in Los Angeles (Preprint 2383, D-3).

The band-pass vented-box enclosure offers several advantages and one major disadvantage. The major disadvantage is that its frequency response is limited to a relatively narrow range, about one to two octaves. This response range can be extended, of course, by the use of equalization, and this is what is done in the KEF 107. With the full amount of equalization, the effective range of the 107's low end extends from 20 to 150 Hz, a three-octave range.

The advantages of the band-pass style of vented-box enclosure include the following: First, because the sound is radiated only from the port, and not directly from the cone, all forms of distortion are potentially lower. This is due to the high linearity of the acoustic resonance system as compared to the mechanical resonance system of the cone driver. Second, the acoustic resonant system acts as an acoustic low-pass filter which attenuates any extraneous noises, such as distortion generated by the low-frequency driver. Third, the low-frequency response of the system rolls off at only 12 dB per octave, the same as a closed-box system, rather than the faster roll-off of a standard vented box, which is 24 dB per octave. This greatly increases the low-frequency energy below the system's 3-dB down point and also much improves the system's low-frequency transient response. Fourth, the sealed box loading one side of the cone adds additional stiffness to the driver and thus increases the system's subsonic power-handling capability. This overcomes one of the major disadvantages of the standard vented-box enclosure, which essentially unloads the driver at frequencies below the vented-box resonance frequency.

Even without its equalizer, the 107 has commendable frequency response, but the equalizer flattens and extends it further.

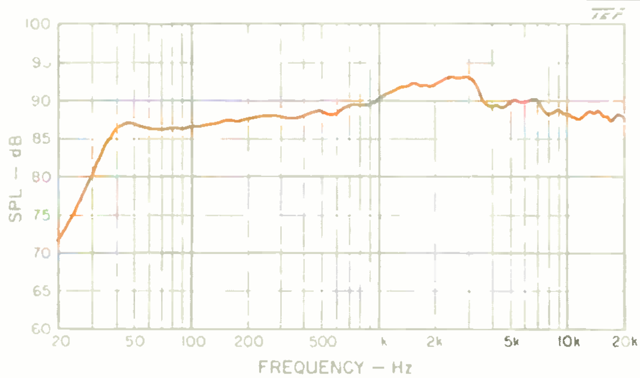


Fig. 2—On-axis frequency response without KUBE equalizer. (0 dB = 20 μPa.)

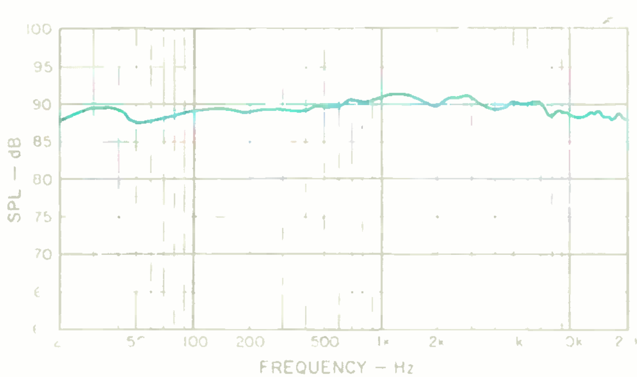


Fig. 3—On-axis frequency response with KUBE equalizer at factory-recommended settings. Note extended bass and flattened midrange.

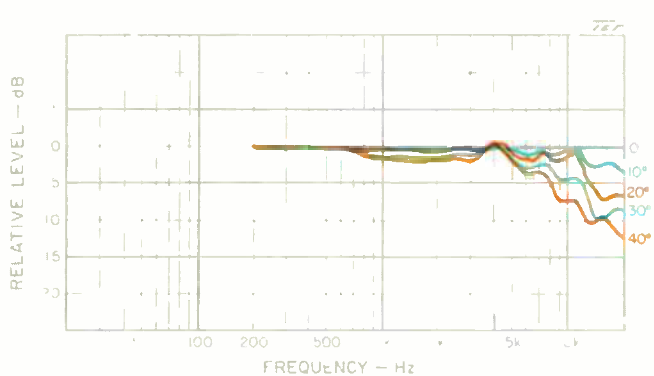


Fig. 4—Horizontal off-axis response, normalized to show deviations from on-axis response.

The KEF 107 actually implements the band-pass vented-box enclosure, using two low-frequency cone drivers. This configuration is shown in Fig. 1. The two drivers feed into a common cavity with a single duct exiting to the outside air. Each driver is loaded by a separate sealed enclosure. The magnet assemblies of the two drivers are connected solidly together with a metal rod; this essentially cancels any inertial forces which could vibrate the cabinet. Additional distortion is cancelled due to the push-pull configuration of the drivers (one driver moves toward its frame while the other moves away).

The duct exit is actually on the top of the low-frequency enclosure, which places it in close proximity to the mid- and high-frequency drivers. Because the duct exit is significantly smaller than an equivalent direct-radiating cone assembly, the total radiating sound area for the whole 107 system is significantly smaller than for an equivalent system with the same acoustic output. The close proximity of low-, mid-, and high-frequency radiating elements in this system ensures more even coverage of the listening area. The frontal area of the system is quite close to that of a typical two-way mini-monitor, but with a vastly higher acoustic output capability that goes down into the response regions of some subwoofers.

The KUBE equalizer is an active, line-level equalizer providing two types of equalization, one fixed, the other variable. The fixed equalization compensates for mid- and high-frequency response errors of the unequalized system and also provides a means for matching the left and right speakers of the stereo pair. The variable equalization allows control of the system's low-frequency output and response shape. Three parameters of the low-frequency response can be controlled: "Extension," which allows setting of the lower cutoff of the system to 50, 35, 25, and 18 Hz; "Q," which allows setting of the low-frequency response shape continuously in the range from 0.3 (overdamped) to 0.7 (maximally flat), and "Contour," which allows setting of the overall low-frequency level below 500 Hz in the range of ±3 dB. These three controls allow a wide adjustment range of the system's low-frequency behavior so that response can be optimized for many different listening environments and types of program material.

The construction and finish of the 107 are of the very best quality. Much attention has been paid to detail, even in normally inaccessible areas. The system can be used without the mid/high-frequency grille assembly without any apologies. The low-frequency and mid/high-frequency enclosures are very solid and essentially inert due to very effective internal bracing and the special configuration of the cabinets. I was not aware of any cabinet wall vibration of any kind during my tests. Connections to the 107 are made via heavy-duty, gold-plated, knurled knob terminals designed to accept large-diameter wire, 4-mm plugs, or spade connectors.

Measurements

Most of the measurements for this review were made with the Techron TEF System 12 analyzer, which uses the technique of Time Delay Spectrometry (TDS) invented by the late Richard Heyser. Mr. Heyser was the senior loudspeaker

The frequency response of the 107 actually achieves the 20 Hz to 20 kHz, ±2 dB, listed in its specs. This is very flat indeed!

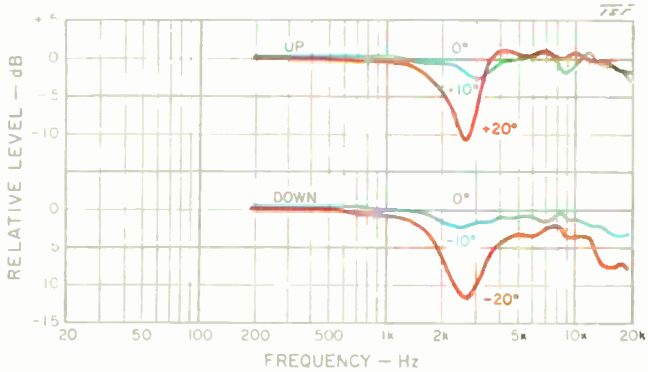


Fig. 5—Normalized vertical off-axis response for angles above the axis (upper curves) and below it (lower curves).

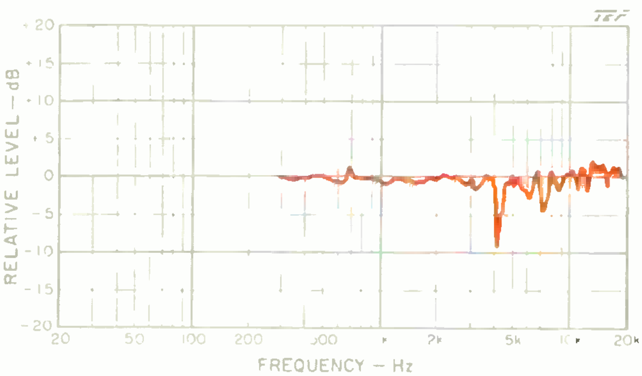


Fig. 6—Effect of grille on on-axis frequency response (normalized).

reviewer for *Audio* for more than a dozen years and used the TDS test technique extensively in his reviews.

TDS allows anechoic chamber-like, reflection-free measurements of parameters such as frequency response, phase response, and time response to be made under decidedly non-anechoic conditions. TDS measurements can be made under high noise-level conditions due to the inherent noise-rejection capabilities of the TDS technique.

The measurements in this review were generally done outdoors, with a combination of elevated free-field measurements, ground-plane measurements, and near-field measurements. Information on the ground-plane technique can be had in Mark Gander's paper in the *Journal of the Audio Engineering Society* ("Ground-Plane Acoustic Measurement of Loudspeaker Systems," Vol. 30, No. 10, p. 723, October 1982). My article on the near-field technique appeared in the April 1974 issue of the same journal ("Low-Frequency Loudspeaker Assessment by Nearfield Sound-Pressure Measurement," Vol. 22, No. 3, p. 154).

The system on-axis frequency response was measured at a distance of 2 meters directly on-axis of the tweeter. The input level was 2.83 V rms, which corresponds to a level of 1 watt into 8 ohms (the system is actually 4 ohms). The on-axis response was corrected to the standard distance of 1 meter for display of the data. The measurement parameters were set so that the data was essentially smoothed with a tenth-octave filter.

Figures 2 and 3, respectively, show the unequalized and equalized response curves of the system. The unequalized response is commendably smooth, exhibiting a lower 3-dB down point at about 35 Hz and actually extending beyond 20 kHz—out to 23 kHz—before dropping rapidly. An upper-midrange peak of about 3 dB, extending from 1 kHz to 3.5 kHz, is also noted. The sensitivity of the system appears to be roughly 89 dB for 2.83 V rms input. Remember that this is a measurement of boundary-free response similar to that measured in an anechoic chamber; the low-to-high frequency balance will change when the system is placed near reflective boundaries in an actual listening situation.

The equalized response curve was run with the factory-recommended equalizer settings of full "Extension" (4), maximum Q (0.7), and a slight "Contour" boost (+1). With these equalizer settings the response actually meets KEF's rating of 20 Hz to 20 kHz, ±2 dB! This is very flat indeed. Note that the equalizer also nicely minimizes the upper-midrange response peak.

The off-axis frequency response was measured and is illustrated in Figs. 4 and 5. Figure 4 shows the normalized horizontal off-axis response plotted on a log frequency scale, but only for angles out to 40° off-axis. Normalization is equivalent to precisely equalizing the on-axis response flat and then measuring the off-axis response. The response does exhibit some high-frequency roll-off above 10 kHz at angles beyond 20°. Figure 5 shows the normalized vertical off-axis response for angles of 10° and 20° above and below the axis. An off-axis crossover dip is evident at about 3 kHz both for 20° up and for 20° down.

For critical listening, the listener should be within a 40° horizontal window centered on the axis of the system, and preferably within a 20° window, for accurate high-frequency response. Fortunately, the mid/high-frequency head module can be freely rotated horizontally to aim it at the preferred listening location.

Figure 6 shows the effect of the grille on the on-axis frequency response. The grille caused some narrow high-Q peaks and dips due to internal reflections in the grille framework. I suggest leaving it off for serious listening; the system does look quite handsome without it.

The on-axis response match between left and right was so close that it approached the repeatability limits of my test setup. The only significant difference occurred in the 9- to 14-kHz range, where the left speaker was slightly louder than the right speaker, about 1.5 dB. I judge this level of right/left matching to be extremely good. Subsequent measurements of the equalizer revealed no differences between its channels.

Figure 7 shows the phase response of the system (equalized) with the delay adjusted for the tweeter. Figure 8 shows the corresponding group-delay measurement. The group delay indicates that the midrange and tweeter arrival times differ by about 0.24 mS (240 µS), with the midrange lagging the tweeter. The increase in delay below 500 Hz reflects the

Handling a kilowatt of amplifier power with ease, the 107 can deliver clean sound at peak levels that could endanger hearing!

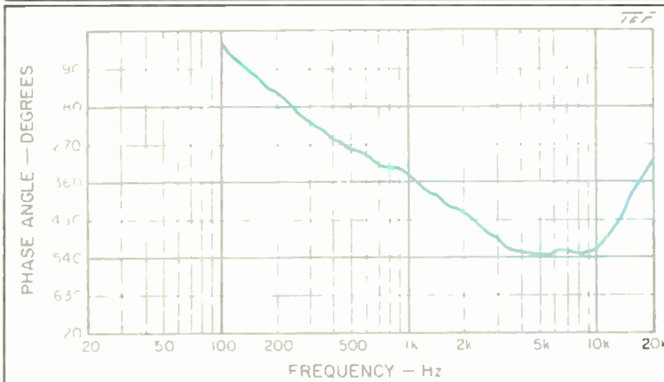


Fig. 7—On-axis phase response of the equalized system, with delay adjusted for the tweeter.

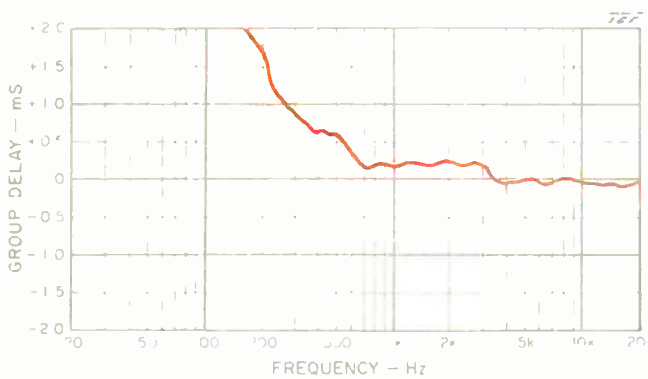


Fig. 8—Group delay corresponding to phase response of Fig. 7.

normal minimum-phase delay caused by the high-pass characteristic of the speaker's response. Most available research indicates that this amount of mid/high-frequency all-pass (flat frequency response) delay is inaudible.

The on-axis energy-time response curve (ETC) of Fig. 9 is shown for a test signal swept over the range from 200 Hz to 10 kHz. In general, the ETC is very well behaved except for a broadening of the response at levels well down from the peak.

The measured impedance between 20 Hz and 20 kHz was so flat and well behaved that no test results need be displayed here. The impedance magnitude varied from a minimum of 3.8 ohms to a maximum of 4.6 ohms. The phase angle was less than 10° for all frequencies below 10 kHz and smoothly rose to roughly +35° at 20 kHz. This means that the load can be considered as a resistor of about 4 ohms, which should be extremely easy for any amplifier to drive. Due to the constant nature of the load, no excessively high transient currents will be required by the load due to reactive effects. For speakers that have widely varying impedance, research has shown that with some program material, drive currents can sometimes reach values two to five times the values computed on the basis of minimum impedance alone. KEF points out that the 107 can actually be considered a typical 8-ohm system insofar as sensitivity and load are concerned.

Figures 10 through 13 show distortion measurements taken on the system. All measurements were made without the equalizer, using near-field techniques. Figure 10 shows harmonic distortion for the single frequency E_1 (41.7 Hz) at increasing power levels. Virtually no higher order harmonics are evident at higher powers. Figure 11 shows the intermodulation data for 250 Hz and 2 kHz. Moderate distortion is shown here because both frequencies are handled by the same driver. Figures 12 and 13 show swept second- and third-harmonic distortion at power levels corresponding to midband levels of 90 and 100 dB SPL at 1 meter. At an axial level of 100 dB in the bass range from 40 to 150 Hz, the distortion was essentially less than 1%. This is commendably low distortion at these levels. For the lower frequencies, the third harmonic predominated, while at higher frequencies, the second harmonic was greater.

To evaluate the maximum peak input and output capabilities of the loudspeaker, I used a peak-power test method similar to that used by KEF to evaluate their professional line of loudspeakers. These tests are very revealing in that they indicate, at each frequency, a system's short-term power-handling capacity and maximum acoustic output capabilities. The method, which uses a shaped tone-burst test signal, is discussed by S. Linkwitz in his *JAES* article in April 1980 (Vol. 28, No. 4, p. 250). This method allows the dynamic peak input and output power capabilities of the system to be evaluated over the complete audio range. The shaped tone burst restricts the energy of the burst to a relatively narrow ⅓-octave bandwidth. The burst lasts for 6.5 cycles and is shaped using the Hanning raised-cosine envelope. This test signal has a good combination of characteristics that include short time duration, restricted frequency range, relatively high peak energy, and a waveform somewhat similar to actual musical waveforms.

The burst is presented at such a low duty cycle that the long-term thermal characteristics of the speaker under test are not exercised at all. In testing the 107, the tone burst was generated at 4 bursts per second, except at lower frequencies, where the repetition rate was lowered so as to maintain a crest factor (ratio of peak to average power) no lower than 20 dB. Note that typical crest factors of music, as recorded on Compact Discs, range from 15 to 25 dB.

The test method consists of evaluating the maximum peak input power-handling capacity and maximum output peak sound pressure levels at all the ⅓-octave center frequencies in the range from 20 Hz to 20 kHz. An extremely powerful amplifier, one which can generate 5,000 peak watts (141 V into a 4-ohm load), was used to drive the loudspeaker under test. (When I first started this project, I thought this amplifier would be plenty powerful enough for any test I would care to make. The measurements that follow showed me how wrong I was!)

The test sequence consisted of determining how much of the special test signal could be handled by the speaker at each frequency before either the output sounded audibly distorted or distressed, or the acoustic output waveform

Band-pass vented enclosures have low distortion and a gentle roll-off, and their frequency limits can be equalized.

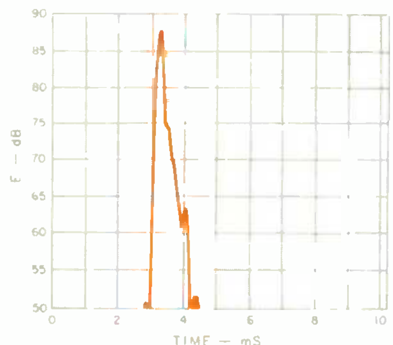


Fig. 9—One-meter on-axis ETC.

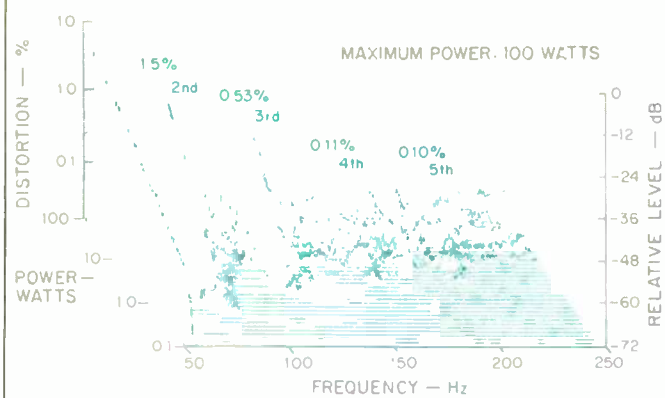


Fig. 10—Harmonic distortion products for a 41.7-Hz tone.

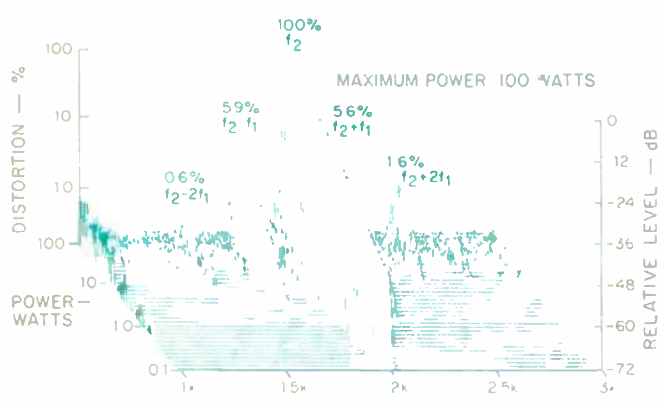


Fig. 11—Intermodulation distortion products for 250-Hz (f_1) and 2-kHz (f_2) tones, mixed 1:1.

appeared distorted on an oscilloscope monitoring the signal coming from a high-intensity microphone (whichever occurred first). At each frequency, the maximum peak input voltage and the corresponding generated peak output sound pressure level at 1 meter were recorded.

Figure 14 shows the maximum peak electrical input power-handling capacity of the loudspeaker. Except for a 2-dB limitation (625 watts) at 300 Hz, the system could handle peak power of 1 kilowatt and greater in the range from 50 Hz to 20 kHz. Above 5 kHz, the system could handle the full 5-kilowatt output of the amplifier (141 V peak)! In this range, amplifier clipping limited testing at higher power. Please realize that, during these tests, the system was producing some very loud sounds! After all this seemingly brutal power testing, the system did not seem to be affected in any way, as far as I could determine

I found that at low frequencies, the 'scope waveform defined the power limit, while at higher frequencies, the audible effects defined the limits. I found that I was quite tolerant of rather high distortion levels (primarily second- and third-harmonic distortion) at low frequencies but was very critical of any slight audible distortion at mid or high frequencies. Note that the 107's band-pass vented-box enclosure, fortunately, filters out most higher order low-frequency harmonics.

Figure 15 illustrates the maximum peak sound pressure levels the system generated at a distance of 1 meter, on axis, for the input levels shown in Fig. 14. Except for a limitation at and near 300 Hz, the system can generate peaks in excess of 115 dB SPL above 50 Hz. Above 1 kHz, the peak levels rise above 120 dB SPL! These levels are loud—hearing protection required!

Also shown in Fig. 15 (dashed curve) are the maximum peak SPLs generated by a pair of 107s in my listening room, as measured at the listening location. The systems could generate peaks in excess of 110 dB SPL at frequencies above 30 Hz (100-dB peaks at 20 Hz). These are wall-shaking levels! The room provides some 5 to 10 dB of low-frequency gain, while a pair of speakers increases the level some 3 to 6 dB.

Please don't interpret this maximum input and output test data to mean that you should run right out and purchase a 2.5-kilowatt power amplifier with a peak power rating of 5 kilowatts to use with your KEF 107s. An amplifier of this size has the capability of blowing up any domestic loudspeaker if continuous long-term power in this range is presented to the system. What I am suggesting is that maybe we need an amplifier with an honest continuous rating of 100 to 200 watts and a peak power rating of 1 to 5 kilowatts (a dynamic headroom rating of 10 to 20 dB). This would allow you to play CDs with typical crest factors of 15 to 20 dB, without clipping, and to exercise the peak sound levels the system is capable of generating—your ears willing!

I measured a wide variety of equalizer frequency responses at different control settings but have elected not to show them here because of space considerations as well as the complex nature of their interactions. Figure 16 shows the maximum boost and cut capabilities of the equalizer from 2 Hz to 1 kHz. Note the very high gain (20 to 25 dB) in the infrasonic range from 2 to 15 Hz. This could potentially

Not many speakers can do justice to everything from gut-thumping heavy metal to chamber music, with no feeling of compromise.

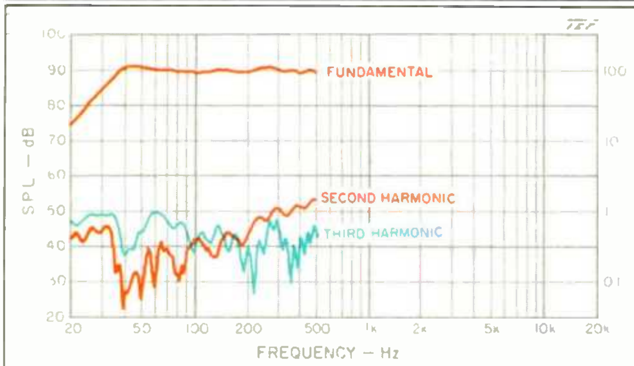


Fig. 12—Second- and third-harmonic distortion level vs. frequency for 90 dB SPL output at 1 meter on axis; input signal is 3.2 V rms (2.5 watts).

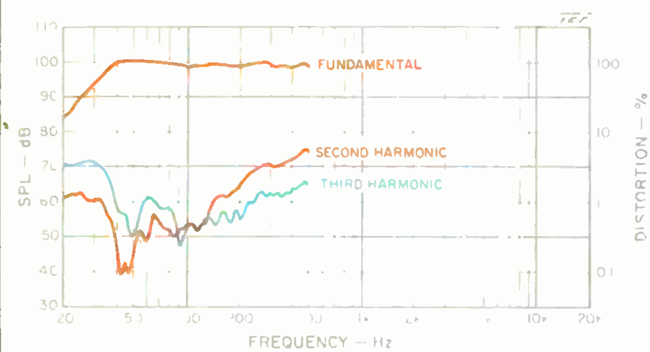


Fig. 13—Same as Fig. 12 but for 100 dB SPL at 1 meter on axis, using input signal of 10 V rms (25 watts).

cause amplifier-speaker overload problems and decreased headroom if any significant infrasonic program energy exists. Lower frequency measurements indicate a first-order roll-off of 6 dB per octave below about 1 Hz. The first production release of the KUBE was actually d.c.-coupled and exhibited problems with some power amplifiers. The current version is a.c.-coupled to minimize these problems.

One source of potential infrasonic energy is from turntable rumble, where tonearm resonances may generate frequencies in the 5- to 8-Hz range. This problem is compounded by the fact that previous measurements showed the speaker's maximum input level below 20 Hz to be only about 6 V rms before excessive distortion is generated. Since I have designed speaker systems, I can sympathize with some of the design goals that led to this high amount of gain, but at least an optional high-pass filter with a high roll-off rate could have been provided to roll off the subsonic frequencies when needed.

I measured the equalizer's maximum input and output voltage capabilities when its controls were set to their maximum boost settings ("Extension" = 4, Q = 0.7, "Contour" = +3). A standard 10-kilohm load was used for these measurements. Maximum voltage was considered to be the point where distortion of the output waveform became visible on an oscilloscope. The measurement (not shown) indicates that the lowest pass-band maximum output voltage was 3.5 V rms at about 2.5 kHz. Make sure you have your power-amplifier gain set high enough so that this limit is not reached by the equalizer. The measurement of maximum input voltage (also not shown) indicates a very low input limit of roughly 0.5 V rms for frequencies below about 15 Hz. This low level is due to the high gain of the equalizer in this frequency range. If you use the maximum gain settings, be aware that you risk overloading the equalizer (and the speaker!) if levels above these limits are sent to the unit. A typical Compact Disc player can generate levels of 2 V rms down to below 2 Hz, so watch out!

Use and Listening Tests

Listening tests were conducted primarily in my basement listening room. The room is somewhat small, with a volume of about 1,500 cubic feet (43 cubic meters). The walls are all non-parallel (by accident, not by design). The systems were placed 20 inches (0.5 meter) away from the wall and separated by about 8 feet (2.5 meters). The upper-frequency modules were aimed at the listening position on the couch along the opposite wall. My ears were at the same height as the tweeters when I was seated on the couch.

A good portion of the listening tests was actually done without the equalizer, not because I have an aversion to equalizers but only because the system sounded quite good without it. The upper-midrange peak at 2 kHz is quite moderate and was audible only when the program material contained energy in its range. I had to listen critically to the midrange frequencies to hear the effect of the equalizer being switched in and out.

It is always a good exercise to listen to a specific amount of spectral aberration being switched in and out of a signal path, to calibrate your ears to the audible effects of certain spectral shifts. The KEF equalizer is quite good for providing specific low-frequency adjustments to allow you to listen to the effects. It's quite easy, for example, to change the Q of the low-frequency roll-off from 0.3 to 0.7 to see how audible the change is. In most situations, the effect is quite subtle.

Several times, I found myself going up to the equalizer and twisting the controls back and forth to see if it was operating, and then doing a double take because it didn't seem to make much difference. The program material must have frequency content in the equalizer's adjustment range for you to hear the equalizer's effect.

Most of my equalizer-in-the-circuit listening was done with the factory-recommended settings of "Extension" = 4 (flat to 20 Hz), Q = 0.7, and "Contour" = +1. With the equalizer in the signal path, the system provided a very neutral listening environment, with no emphasis or de-emphasis of any part of the frequency range. Imaging was very stable and consistent. Reproduction of male singing voice with acoustic guitar was very accurate and realistic. Female vocals showed no hint of spectral imbalance. Because of the very close right/left matching of the speakers, there was absolutely no lateral shift of image with changing frequency. I was aware of some moderate roll-off of the extreme high

Reasonable size, high output and power-handling capacity, and smooth response make the 107 a true reference standard.

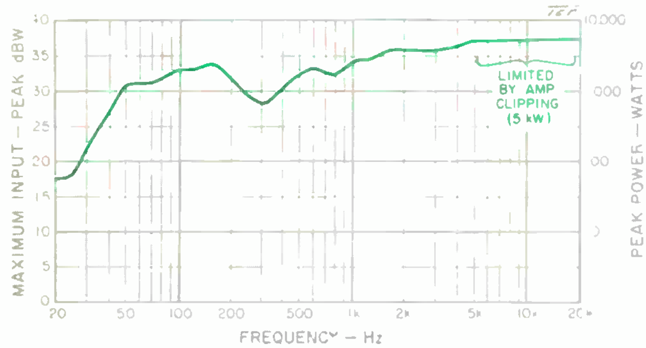


Fig. 14—Maximum peak input power for moderately clean output (see text). Above about 5 kHz, note that limitations are imposed by clipping of a 5-kilowatt amplifier, not by the speaker.

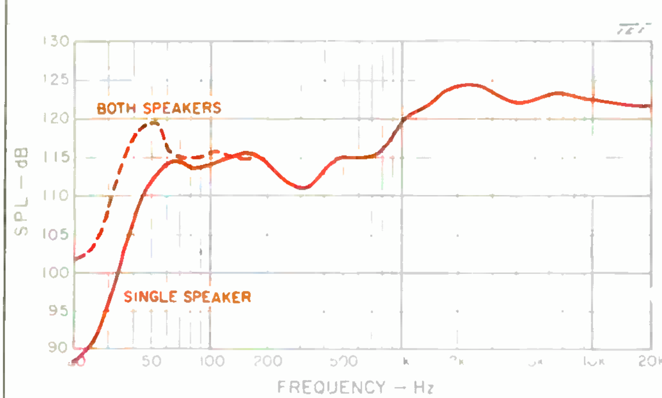


Fig. 15—Solid curve: Maximum peak SPL output at 1 meter, on axis, for input levels shown in Fig. 14. Dashed curve: Combined output of two 107s measured at the listening position, including low-frequency gain from room interactions.

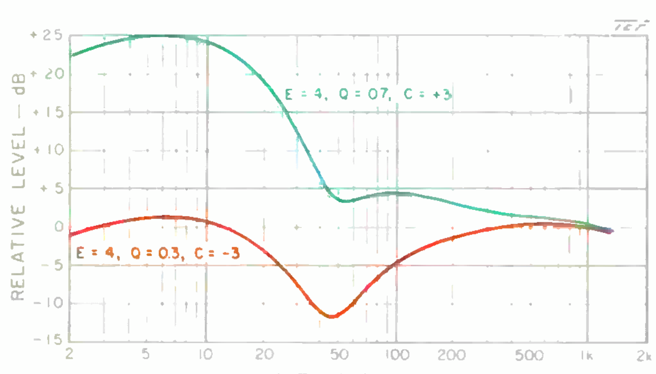


Fig. 16—Maximum boost and cut capabilities of KUBE equalizer supplied with the 107. Note the high gain in the infrasonic region, below 15 Hz (see text).

frequencies at relatively large off-axis angles. Make sure that the high-frequency modules are aimed at the listening position to minimize this effect.

I initially was somewhat apprehensive about using the equalizer at settings near full boost, because of the amount of boost I knew it could provide. Subsequent listening proved that my worries were unfounded, however. At only one time was I aware of any overload problems I could attribute directly to the equalizer: The cannon shots on Telarc's CD of Tchaikovsky's "1812 Overture."

I suggest you make sure your power amp's input gain settings are set high so that the drive level to the equalizer is kept low. If you use the equalizer in the tape loop of a receiver, you could be in trouble because the full signal at the AUX input is typically fed through to the tape out, with no attenuation. This means that the full 2 V rms output (at 0-dB record level) of your CD player may be applied directly to the equalizer. If your player has adjustable outputs, use them to reduce the drive level to the equalizer.

The low-frequency response was always quite solid and very much in evidence when the program material demanded it. Pipe-organ pedal notes were reproduced with much authority. The low frequencies could get up to very impressive sound levels without any audible stress whatsoever. On transient bass passages, such as rock kick drum, the lows were very tight and could be turned up to levels that would make a professional drummer happy. The extended low-frequency response of the system unearthed some very low-frequency rumble, of which I was previously not aware, on a couple of my classical Compact Discs.

With three teenagers in my house, our listening system is subjected to many varied types of program material, from Mozart and Dave Grusin to ZZ Top and Run-D.M.C. The KEFs proved themselves very versatile in realistically reproducing everything from the subtleties of chamber music and delicate female vocals, played at low to moderate levels, to heavy-metal rock music, played at gut-thumping levels that only my teenage son could appreciate.

There are not too many systems that can do justice to such extremes in program material and not come out as a middle-of-the-road compromise. The KEFs did an extremely good job on all types of program material. The 107s are a fine example of a system that provides a good combination of physical size, high maximum output capability, very smooth response, and high power-handling capacity. In the true sense of the term, the KEF 107s can be considered a reference standard.

D. B. Keele, Jr.

D. B. Keele, Jr. is Manager of Software Development for the Techron Industrial Products Division of Crown International, Inc., makers of TEF measurement equipment on which the tests for this review were performed. Don Keele is a Fellow of the Audio Engineering Society, so honored for his work on vented box design, and helped develop the near-field woofer measurement technique. He has also worked at Electro-Voice, Klipsch, and JBL. He says that he is most proud of his design work on constant-directivity horns, an area where he holds three patents.

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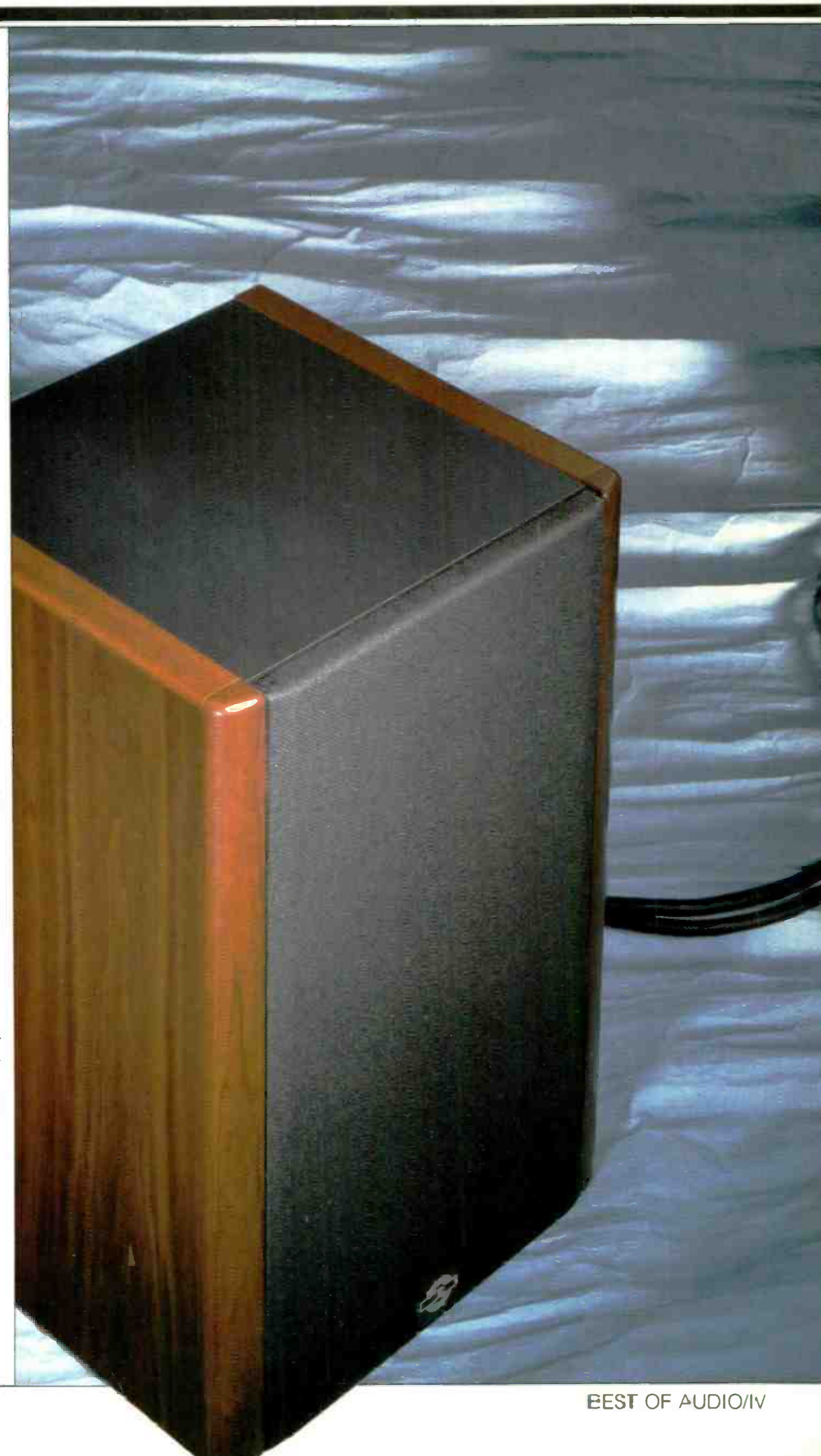

MB QUART 280 SPEAKER

Manufacturer's Specifications

System Type: Two-way, sealed-box woofer.

Drivers: 8-in. (20.5-cm) woofer and 1-in. (2.5-cm) dome tweeter.

Crossover: 1.5 kHz.

Frequency Range: 40 Hz to 32 kHz.

Sensitivity: 88 dB SPL at 1 watt/1 meter.

Nominal Impedance: 4 ohms.

DIN Power Handling: 80 watts.

Music Power Handling: 100 watts.

Dimensions: 10½ in. W × 17⅜ in. H × 11¼ in. D (26.8 cm × 44.1 cm × 28.6 cm).

Weight: 21.2 lbs. (9.6 kg).

Price: $599 per pair.

Company Address: MB Quart Electronics, 25 Walpole Park South, Walpole, Mass. 02081.

(Originally published December 1988)

Photograph: ©1988, Bill Kouirinis

The MB Quart 280 has to be a special two-way bookshelf speaker just to survive the competition. Loudspeakers utilizing an 8-inch woofer and a 1-inch dome tweeter, and slight variations, are the most common types available to the audio enthusiast. Prices range from under $200 to $4,600 per pair. At $599 per pair, the MB Quart 280 is right in the thick of this saturated market. However, the 280 is imported from Germany and has a multitude of features that sets it apart.

The West German parent company, MB Quart, is experienced. They began in 1963 as a supplier of drivers to other speaker manufacturers. For the past six years, they have successfully marketed systems under their own name throughout Europe. Now they are expanding to North America with the introduction of a line of five speaker systems which are said to have had their sound adjusted to suit American tastes.

Personally, I have always felt that "correct is correct" and that "taste" was best handled by tone controls. How can the European and the American 280s both be correct? Alex Goetzenberger, a company spokesman, explained that Europeans tend to listen in an analytic manner and for shorter periods of time. He said that differences in preference are becoming less pronounced, but Americans tend to have the music playing all the time while they are involved in other activities. Thus, a slightly more prominent midrange and treble spectrum is offered in the European speakers and more bass capability in the U.S.

Quality materials and cabinet construction help differentiate the 280 from many other speaker systems. The particleboard panels, an unusual ⅞ inch thick, are manufactured specifically for MB Quart speaker cabinets. They are composed of five layers of differing density and particle size, to achieve strength and damping. The boards are assembled with tongue-and-groove joints and thick wood edge trim. Extra-thick veneer on the sides ensures that minor scratches can be repaired without sanding, and the front panel has a velvet-like, black "flocked" treatment that is electrostatically applied and is claimed to control unwanted diffraction. The rear, top, and bottom panels are brown plastic laminate with an attractive contrasting texture. Knit fabric, in a color matching the speaker's finish, is stretched over a simple particleboard frame to form the removable front panel. Heavy-duty five-way binding posts are partially recessed in the rear panel; their spacing does not allow the use of double-banana plugs.

The titanium dome tweeter, operating above 1.5 kHz, is the most intriguing technical component. Titanium requires expensive processing to form it into a thin dome shape, but its combination of strength and lightness makes it worth the trouble: The first breakup modes can be pushed to well above 20 kHz. (It is interesting to note that the seemingly opposite approach, used in "soft dome" tweeters, can also result in good sound. Highly damped fabric dome tweeters have "benign" breakup modes beginning very low in their frequency range.)

The MB Quart 280's crossover frequency of 1.5 kHz is an octave lower than that typically found in two-way systems. This crossover brings in the tweeter's wide directivity just as the woofer pattern is narrowing down. The problem is to

The 280's directivity is broad in both planes and avoids the narrowing that many two-way loudspeakers exhibit.

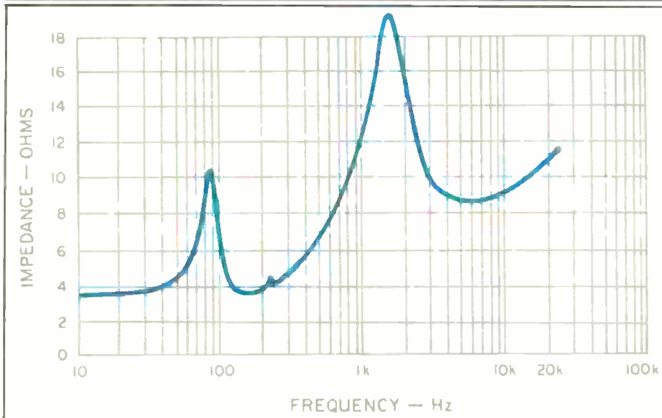

Fig. 1—Magnitude of impedance.

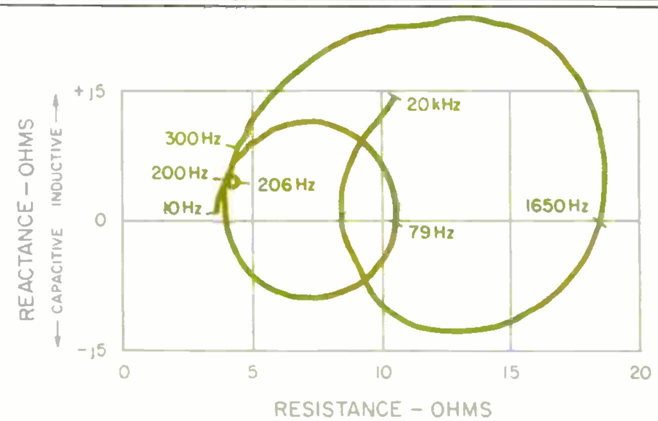

Fig. 2—Complex impedance. Note the slight glitch at about 200 Hz; see text.

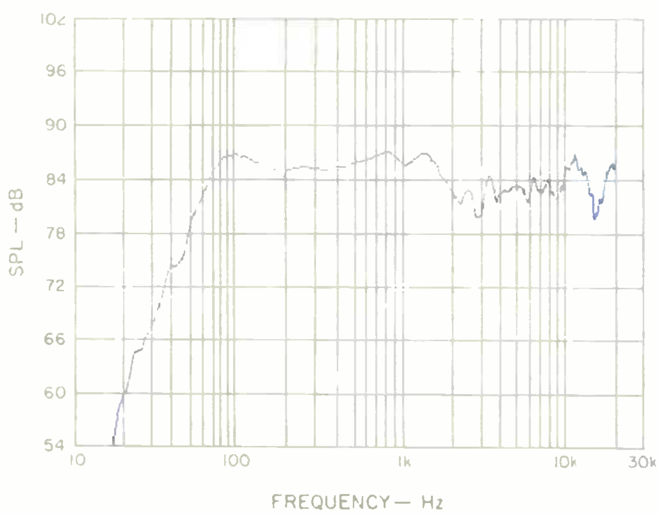

Fig. 3—One-meter on-axis anechoic frequency response, with an input of 1.0 watt into 4 ohms (2.0 V).

design a tweeter to go that low with good efficiency, low distortion, and reliability.

The 280's tweeter starts with a large magnetic system for efficiency and linearity. An additional ferrite doughnut is glued to the rear of the tweeter, in opposing magnetic polarity, to force flux to stay in the circuit, further increasing efficiency. Voice-coil heat is conducted across the gap to the magnetic circuit by Ferrofluid, which also helps damp the primary mass/suspension resonance. Passages coupling the air volume under the dome with the volume under the surround help to achieve the low resonance frequency needed to respond down to 1.5 kHz. Essentially, the dome gets a larger enclosure.

High-quality components are used in the unit's crossover, and heavy wire runs to the speakers. Crossover slopes are nominally 12 dB per octave.

The woofer has about 13% more surface area than a conventional 8-inch device. This is accomplished within an 8-inch frame width by using a narrow flange with four protruding mounting ears. Other than that, the 280's stamped-frame, paper-cone woofer is not unusual, except for its longer-than-average linear travel.

Unusually good imaging is claimed for this loudspeaker, and it is attributed to wide directivity and diffraction control. Wide directivity is achieved by using the low 1.5-kHz crossover frequency and a "dispersion ring" in front of the tweeter dome. Diffraction, the re-radiation of sound as it passes over a sharp discontinuity, is claimed to be reduced by flush tweeter-mounting screws and the flocked panel treatment. The fact that the substantial wood frame and woofer protrusions would likely produce strong diffraction is claimed to be handled by the asymmetrical tweeter mounting. Diffraction is controlled—not necessarily eliminated.

Measurements

Input impedance is shown as a plot of magnitude versus frequency in Fig. 1 and as reactance versus resistance in Fig. 2. Both plots show fairly typical performance for a 4-ohm-rated speaker except for the curious little glitch at just over 200 Hz. To investigate this, I applied a 206-Hz sine wave at about 10 watts. I discovered that one edge of the woofer frame was vibrating heavily. Pressing on it with the blade of a screwdriver stopped the vibration and increased the sound output. I removed the woofer and discovered that the foam gasket between the driver and the box had slipped out of place. With the gasket back in its proper position, the glitch disappeared, but some vibration remained in the speaker frame.

Figure 3 is the on-axis frequency response measured at 1 meter without the influence of room reflections. Sensitivity is about 85 dB below 1.5 kHz with 2 V input. MB Quart's claim of 88-dB sensitivity appears to be made using 2.83 V input,

The 280's time response is about as close to ideal as you are likely to get from a loudspeaker.

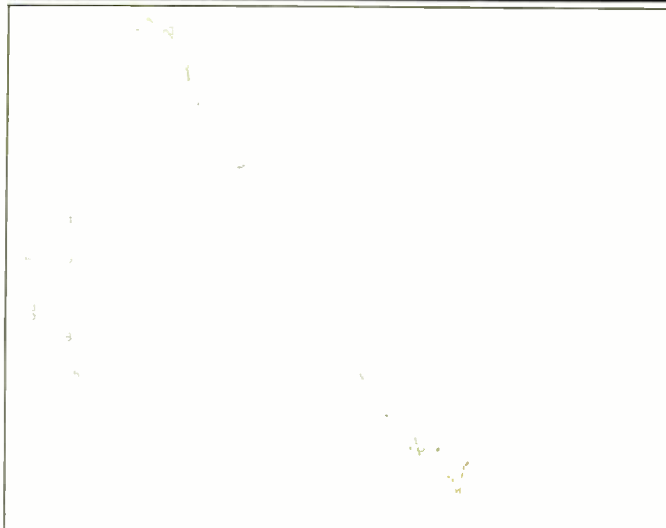

Fig. 4—One-meter on-axis anechoic phase response.

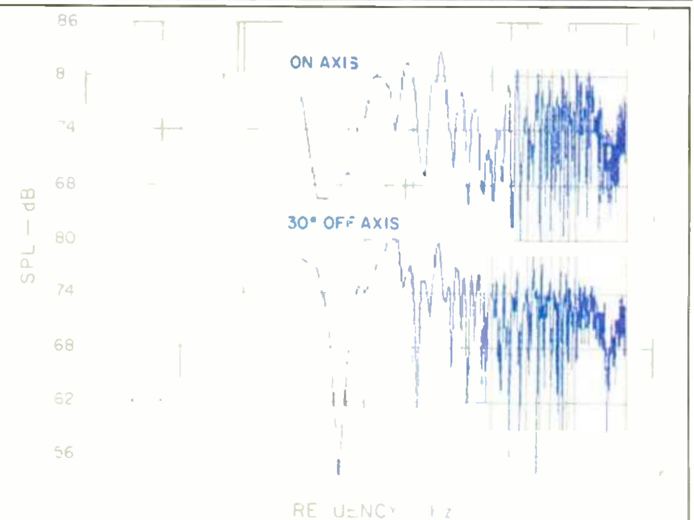

Fig. 5—Three-meter room response measured on-axis and 30° off-axis; for clarity, off-axis curve has been lowered.

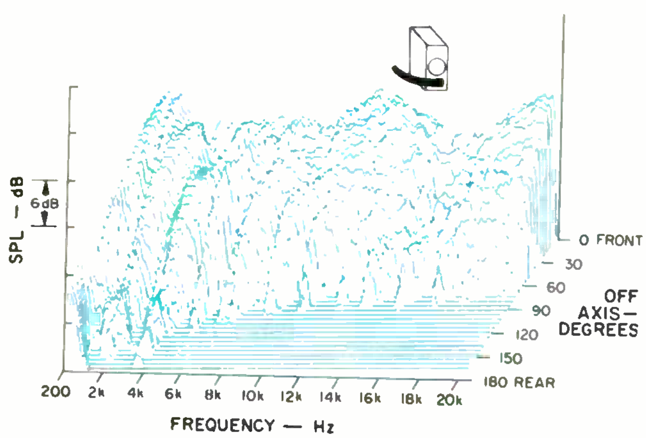

Fig. 6—Horizontal off-axis responses taken from the front, around the side, to the rear of the speaker.

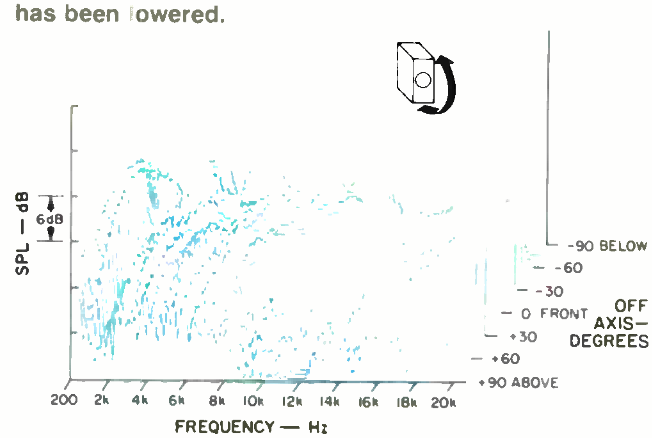

Fig. 7—Vertical off-axis responses taken from below, up the front, to the top of the speaker.

which is equivalent to 1 watt into an 8-ohm speaker. The 280, however, is a 4-ohm unit, but using the same drive levels, the measurements agree. Look at the power your amplifier will provide into a 4-ohm load and consider the 85-dB figure. This sensitivity is a bit on the low side, but reasonable. An amplifier capable of 100 watts per channel should produce peaks around 108 dB SPL in a typical room.

Low-frequency response shows the roll-off of 12 dB per octave expected from a sealed-box system below its resonance frequency, which is 79 Hz for the MB Quart 280. Apparently, this system is intentionally a little underdamped, which gives a slight boost at 100 Hz and extends the −3 dB frequency to about 60 Hz. For the same cabinet size, designing for higher sensitivity would have the indirect effect of increasing damping and raising the cutoff frequency. The upper range is reasonably smooth but is suppressed about 3 dB above 1.5 kHz. (Could this be the tweak to Americanize the 280? I think the subject is a little more complicated, and I will go into it further when we look at the directivity plots.) The titanium tweeter shows no sign of giving up by the end of the plot at 20 kHz.

Phase response (Fig. 4) is just what one would expect of a classically designed two-way loudspeaker with a second-order crossover. Phase lead is expected at very low frequencies, where the response is rising. Phase passes through −180° in the midrange, due to the crossover, and lags a further 180° due to the eventual tweeter roll-off. Each speaker in this pair matches the other quite closely.

Though placement and aiming were noncritical, bass and spaciousness where best with the 280s out from the walls and off the floor.

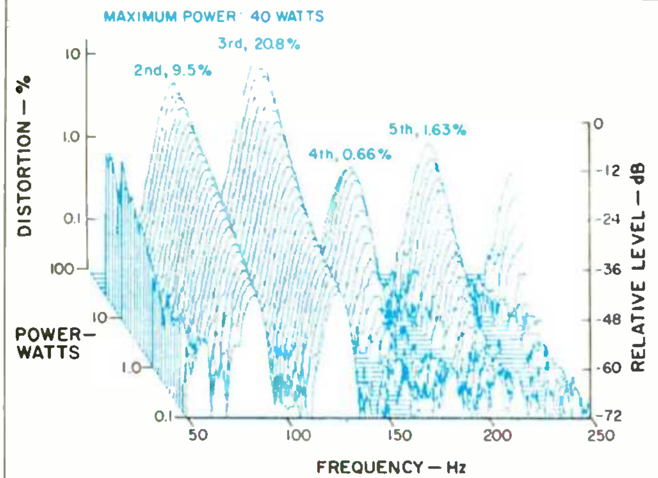

Fig. 8—Harmonic distortion products for the tone E_1 (41.2 Hz).

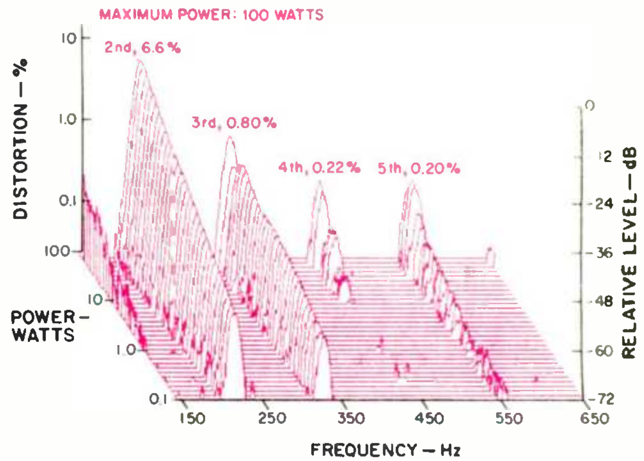

Fig. 9—Harmonic distortion products for the tone A_2 (110 Hz).

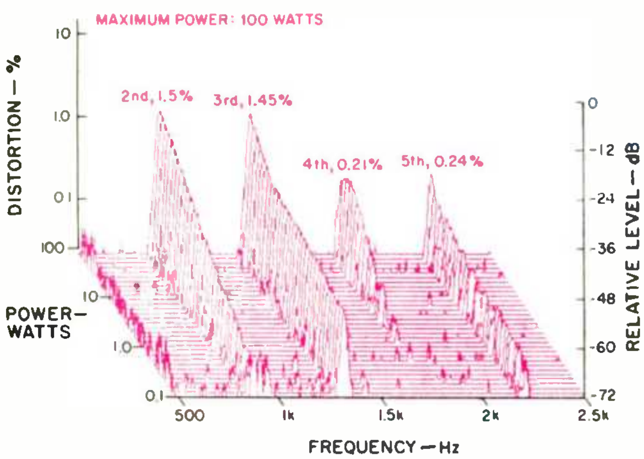

Fig. 10—Harmonic distortion products for the tone A_4 (440 Hz).

The on-axis and 30° off-axis 3-meter room response (Fig. 5) includes the important early reflections that might be found in any listening room if the speakers are positioned as I used them. The locations which worked best for me in the listening test were 19 inches off the floor, 48 inches away from the side wall, and 36 inches in from the wall behind the speakers. The plots in Fig. 5 show reduced output at 300 Hz, which is traceable to sound bouncing off the floor on its way to the microphone. The longer path delays the signal such that 300 Hz suffers a partial cancellation. Above 300 Hz, the average response is uniform and extended, even at 30° off-axis.

Horizontal directivity versus frequency is shown in Fig. 6, a "3-D" plot. It is wide at low frequencies, as expected from a physically small loudspeaker. Directivity gradually narrows at high frequencies but never becomes overly directional. Something unusual to note is that there is no evidence of the narrowing directivity from 1 to 3 kHz found in most two-way speaker systems. I would consider this a technical plus for the MB Quart 280, but I have heard many fine systems that do exhibit midrange narrowing. Perhaps the 280's on-axis response in this range must be suppressed—as we observed it to be in Fig. 3—in order to reduce total input to the room. I do not want to criticize this unit for being "too good"; rather, I am wondering how other speakers that do not measure as well on this can *sound* so good.

Vertical off-axis plots—taken from below, up the front, to above the speaker—are shown in Fig. 7. The same wide directivity is evident, and there is the expected interference between the two drivers in the crossover range for angles well off-axis.

Linearity of the MB Quart 280 was measured in three different ways. First, the plots of Figs. 8, 9, and 10 show the harmonics generated for the musical tones E_1 or 41.2 Hz, A_2 or 110 Hz, and A_4 or 440 Hz in steps from 0.1 to 100 watts. Since the lowest fundamental is below the system's operating range and thus down in level, the harmonics, which are in the speaker's range, appear to be very high. In the earlier listening test, the 280 handled this input in a dignified, if quiet, manner. The plots at higher test frequencies are more reasonable, with 440 Hz showing quite low distortion.

The second linearity test is to observe the effect of a low tone (E_1 or 41.2 Hz) on a high tone (A_4 or 440 Hz). This is the intermodulation test plotted in Fig. 11. Distortion remains below 3% until about 20 watts input. Above this power level, the lower test frequency begins to move the voice-coil out of the magnetic gap twice during each cycle. This produces a cyclic modulation of the 440-Hz tone being measured. A

At $599 per pair, you won't get everything, but the 280 will deliver low coloration, good range and dynamics, and a forward presence.

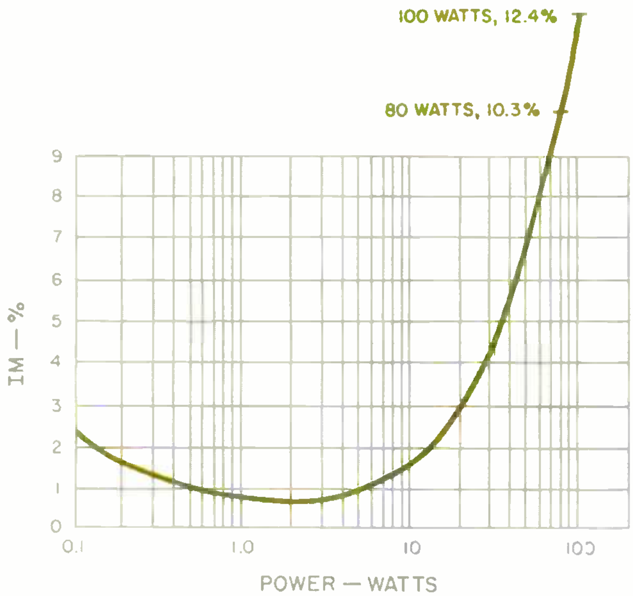

Fig. 11—IM distortion on 440 Hz (A_4) produced by 41.2 Hz (E_1) when mixed in 1-to-1 proportion. Readings below 3 watts are due to noise.

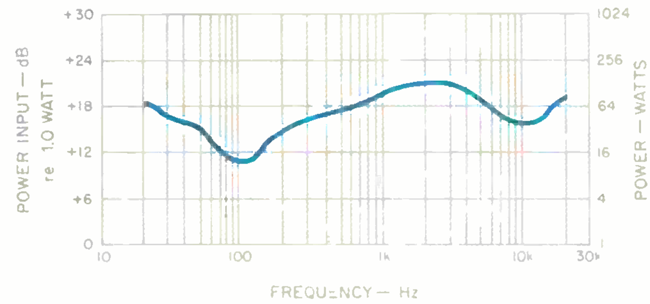

Fig. 12—Power linearity.

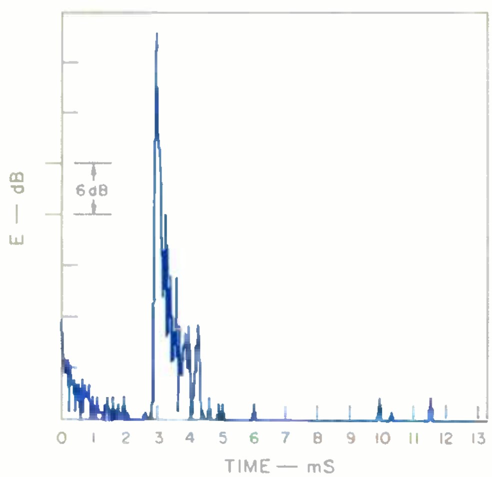

Fig. 13—One-meter on-axis energy vs. time response.

distortion level of 10%, an objectionable amount, is reached at 80 watts.

The third linearity test checks midrange and treble as well as lower frequencies. Increasing power is applied until the acoustic output fails to track the input by 1 dB. Figure 12 plots the power input for this mistracking. It can be seen that it doesn't pay to use ultra-high powered amplifiers with this small loudspeaker.

Figure 13 shows the energy versus time response of the 280. Ideally, the test pulse's energy should be reproduced at one instant in time. The tested unit's response is about as close to this as you are likely to get from a speaker. Obviously, diffraction, which would show up as a widening of the pulse, is not a problem.

Use and Listening Tests

As usual, listening tests were performed before the technical measurements were made. More listening after the testing convinced me that the loose gasket found in one of the speakers had not had a noticeable effect on the sound. I found placement and aiming of the MB Quart 280s to be relatively noncritical. Spaciousness and smoothness of bass response were best, however, with the speakers well out from the walls and 19 inches off the floor. They could be placed next to other small speakers without ill effects, which led me to make lots of comparison tests in different rooms.

In these comparisons, the performance of the 280s was consistently good in all respects, while that of other speakers varied. Few could match the balanced, uncolored character of the 280s. Of those that did, fewer still had as extended a frequency range. The MB Quarts rendered the sound of recordings with a close-up perspective more accurately than their competitors did, and made solo instruments and voices sound like they were forward and present in the room. Some other models, comparable in size and price, sacrificed this presence for an improved sense of depth and spaciousness.

Speakers are the weakest link in the sound-reproduction chain. Frequently, they are also the most visible and the most costly. In the price range of under $600 per pair, you can't get everything from any speaker. You should concentrate on those qualities that mean the most to you. For low coloration, good range and dynamics, and a forward presence, the MB Quart 280 is a top contender. Visual appeal and convenient size complete the package. This loudspeaker is likely to be the best choice for many audiophiles.

David L. Clark

5

FUSELIER 3.8D LOUDSPEAKER

Manufacturer's Specifications

System Type: Three-way; vented woofer.

Drivers: 8½-inch woofer, 2-inch dome midrange, and 1⅛-inch dome tweeter.

Crossover Frequencies: 700 Hz and 6.5 kHz.

Frequency Response: Within ± 1.5 dB of designed spectral balance.

Sensitivity: 88 dB SPL at 1 meter for 2.8 V input.

Impedance: 5 ohms at 350 Hz, 2.2 ohms at 9 kHz.

Recommended Amplifier Power: 20 to 160 watts.

Dimensions: 23½ in. H × 10¾ in. W × 17¾ in. D (59.7 cm × 27.3 cm × 45.1 cm).

Weight: 39 lbs. (17.7 kg).

Price: $2,100 per pair.

Company Address: c/o Audio Products, Inc., 3 Cleveland St., Headland, Ala. 36345.

(Originally published June 1989)

Photograph: David Hamsley

All four of John Fuselier's loudspeaker designs are high-end audiophile systems. High end, however, does not automatically mean high price. Rather, it is a priority which places quality of sound before quantity of sound. The smallest Fuselier, the 2.5, at $775 per pair, is admitted to have low acoustic output, while its sound quality from midbass up is claimed to be excellent. The $2,100 per pair Model 3.8D, reviewed here, is poised at the top of the line, where sonic excellence, wide frequency range, and adequate acoustic output can coexist.

In his quest for quality, John Fuselier's first priority is achieving a very smooth frequency response over a practical range of off-axis listening positions. This requires the best available drivers, precision crossovers, and control of resonance, reflection, and diffraction.

The drivers in the 3.8D are all standard products from the highly respected Danish manufacturer, Dynaudio. The woofer, midrange, and tweeter are Models 21W54, D-52AF, and D-28AF, respectively. Fuselier prefers selected off-the-shelf components because they will be available in the future should repair be necessary. The retail price of these basic components is a hefty $250, and I don't suggest trying to "roll your own," because there is a lot more here than three speakers in a box.

A complex crossover network is used in the three-way Fuselier 3.8D. Next to driver selection, this circuitry is the most important factor in achieving smooth frequency response. There are three main design considerations. The first of these is selecting a circuit configuration and component values which will complement each driver to achieve the best acoustic bandpass responses. This can be considered a form of fixed equalization. Second, the acoustic phase shifts of adjacent high- and low-pass filters should be made to track each other, preventing polar-pattern shifts through the crossover range. Finally, when all filters are combined, the input impedance must present a reasonable load to the power amplifier.

Crossover components in the 3.8D are mounted to a p.c. board inside the cabinet. I counted eight film-type capacitors, two nonpolar electrolytics, and four air-core inductors. I believe there are more components under the board. All connections are soldered, which, in my experience, results in greater reliability than push-on connectors crimped to the wire. The tight-tolerance components help match the two loudspeakers of the pair for best imaging.

A stepped-front cabinet design is used to correct for the different acoustic centers of the drivers. The depth of the woofer cone places the voice-coil and the acoustic center of the woofer well behind its mounting flange. With a conventional flat-baffle mounting, the woofer's acoustic center would be behind that of the midrange dome. This is avoided by a 49-mm step in the front panel, which moves the woofer forward. The midrange flange overlaps that of the tweeter, moving the midrange slightly forward as well. This overlap also serves to position the domes as closely together as possible, keeping them in phase over a greater vertical radiation angle. Fuselier claims smooth response over a ±10° angle, which is quite good. A 45° bevel board is used to smooth out the step between woofer and midrange. I would expect this to cause an interfering reflection in midrange radiation, but leaving the discontinuity presumably would cause even more problems.

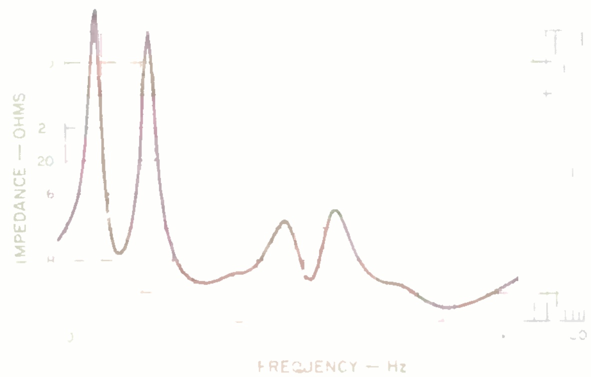

Fig. 1—Magnitude of impedance. Note the dips at the vent tuning frequency (27 Hz), the woofer/midrange crossover frequency (700 Hz), and where the tweeter comes in (10 kHz).

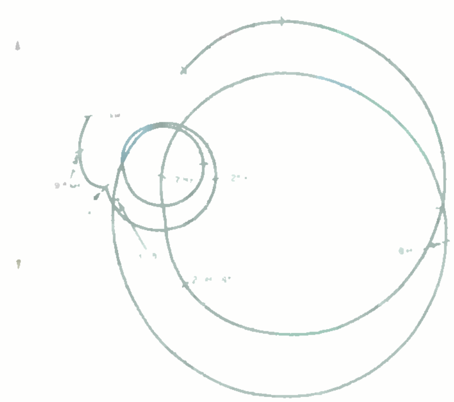
Fig. 2—Complex impedance, showing reactance and resistance vs. frequency.

Mechanical and acoustic cabinet resonances are controlled by two angled, internal, cross-bracing baffles. These plates acoustically separate the parallel surfaces of the enclosure, reducing simple resonant modes. One-inch foam

Positioning is not very critical, which means that suboptimal placement does not greatly degrade the 3.8D's sound reproduction.

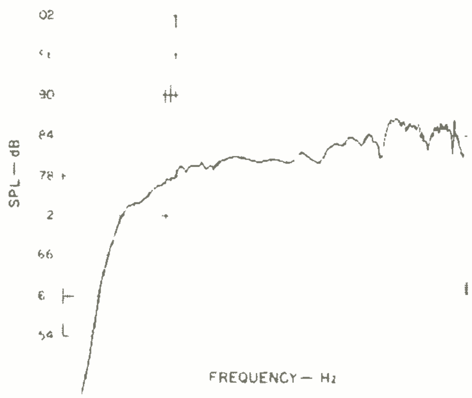

Fig. 3—One-meter on-axis anechoic frequency response, with an input of 1.0 watt into 4 ohms (2.0 V).

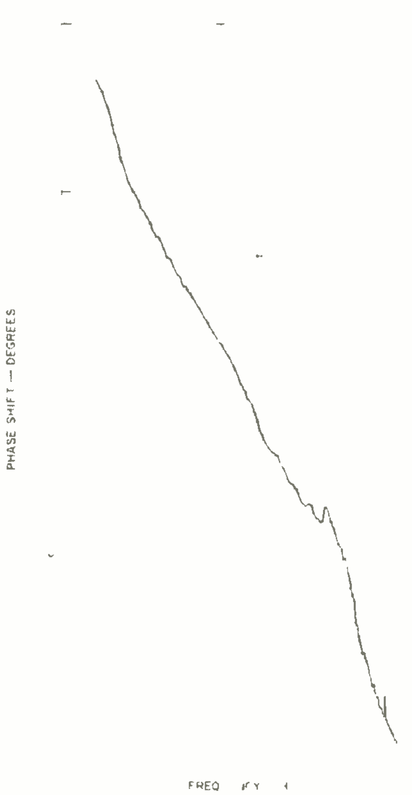

Fig. 4—One-meter on-axis anechoic phase response. The large amount of phase shift shown is not unusual in three-way systems.

blankets line most of the interior, to absorb the woofer's rear radiation. This absorption also reduces low bass output. Fuselier's priorities show here: Bass quantity is sacrificed for improved resonance control.

Other Fuselier loudspeakers use circuit breakers for protection, but the 3.8D is unprotected. Fuselier is justifiably concerned with the breaker's contact integrity after it has cycled a few times. The 3.8D is no more prone to burnout than any other unprotected speaker, and the manual states that distortion will be heard before the system is damaged.

The cabinet finish of the 3.8D is superb. The wood veneer on top, bottom, and sides is sealed, stained, and hand-rubbed with tung oil, giving a deep, glossy finish. The front and rear are painted black, and an elaborate wood grille frame allows brown stretch fabric to cover the protruding drivers without restricting side radiation. Nevertheless, Fuselier claims the speakers are calibrated with the grilles off, implying that this is how the serious listener will use them.

There are no controls on the 3.8D, just two input terminals flush-mounted on the rear. Either double-banana or specialty connectors can be used to connect heavy speaker cable.

Measurements

All measurements were made with the grille removed and, as with the listening tests, were conducted using a solid-state amplifier of low output impedance, as recommended, to achieve the designed spectral balance.

The magnitude of input impedance versus frequency is shown in Fig. 1. At low frequencies, dual peaks straddle the vent tuning frequency of 27 Hz. This vent frequency is one of the more useful pieces of information on this plot, because it is an indicator of the lowest usable frequency of a vented system. At this frequency, the required cone excursion is very low, but it must increase slightly at frequencies above and a great deal below this point. Crossover-related impedance bumps occur around the 700-Hz crossover frequency, and a low of 2.2 ohms is reached at about 10 kHz, where the tweeter comes in.

It is because of this dip to 2.2 ohms that Fuselier recommends a low output-impedance solid-state amp. Tube amps are usually higher in output impedance and could be pulled down 1 dB around 10 kHz, relative to the midrange. You might prefer this reduction, but it is just an expensive way to equalize. Speaker-wire resistance will cause a similar problem; a 15-foot length of 18-gauge zip cord will also drop the highs 1 dB or so. The low impedance will not cause a modern, 4-ohm-rated amplifier to lose output or distort audibly, but the amp will be working hard in this frequency range. Amplifier overheating won't be a problem because high-level 10-kHz signals are infrequent in music. Needless to say, use heavy wire and don't parallel another set of 3.8Ds for music on the patio.

Reactive and resistive components of the input impedance are plotted in Fig. 2, with frequencies and phase angles of interest called out. At frequencies where there is a large reactive component and a small resistive component, the amplifier has to absorb electrical energy from the loudspeaker over part of the cycle. This condition, also indicated by a large phase angle, is more difficult than delivering energy to the speaker over the entire cycle. The 3.8D is a

The spectral uniformity of the 3.8D was outstanding, with low coloration and good overall balance.

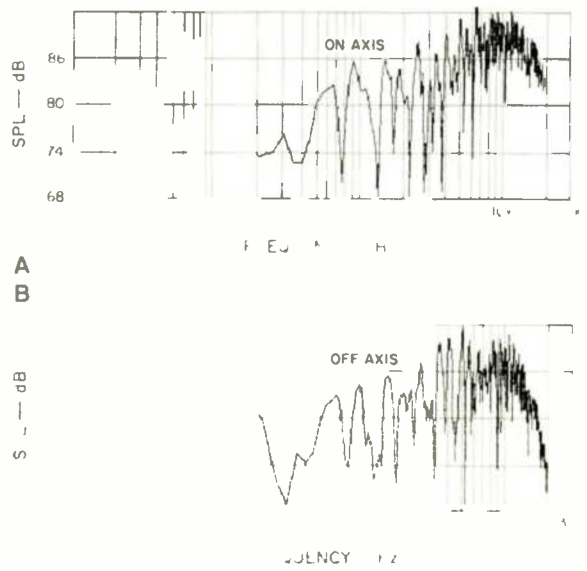

Fig. 5—Three-meter room response, measured on-axis (A) and 30° off-axis (B). Note the smoothness of the off-axis curve.

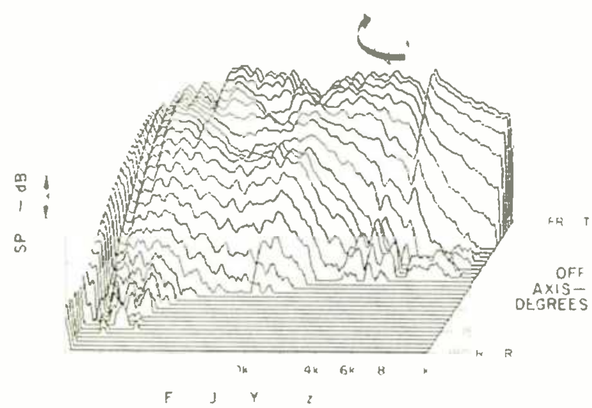

Fig. 6—Horizontal off-axis response plots taken from the front, around the side, and to the rear; see text.

fairly difficult load, but it will not activate the protection circuitry of common, modern "mid-fi" or better amplifiers.

The 1-meter on-axis frequency response curve of Fig. 3 correlates strongly with what I heard. The 3.8D has a generally smooth response that rises gently with increasing frequency. Bass response rises slightly from its 30-Hz cutoff to 100 Hz. Midrange and high-frequency smoothness is marred only by the slight dips at 4 and around 9 kHz. Experimentation confirmed that the 4-kHz dip is due to a reflection from the angled filler panel; the 9-kHz dip is caused by interference between midrange and tweeter radiation. Both of these micro-sized problems go away when the response is measured slightly above the front-panel axis, where Fuselier might reasonably expect the listener to be.

The phase shift accompanying the amplitude response of Fig. 3 is shown in Fig. 4. This large amount is not unusual in a well-designed three-way system. It is more important that the phase shifts of the individual drivers track each other and, of course that the stereo pair is matched. The 3.8Ds are very well behaved in these ways.

The test for 3-meter room response measures frequency response as affected by the early reflections found in a typical room. For these measurements, the speakers were placed in positions previously used in the listening evaluation, with the front of the cabinet 3 feet from the wall behind the speaker and 4 feet from the side wall. The speakers were angled 30° inward and mounted 24 inches off the floor, on stands. The on-axis plot (Fig. 5A) represents what a listener near the center of the room would hear. Its average is smooth but even brighter than shown in the anechoic curves. At 30° off-axis (Fig. 5B), the still-smooth average indicates wide directivity and good room interaction. The usual floor-bounce interference of raised speakers notches the range around 300 Hz.

Amplitude response off-axis is measured every 6° over a 180° degree range and is plotted in a "3-D" format. This is shown in Fig. 6 for the horizontal plane, starting in front and going around the side to the rear. None of the midrange narrowing found in most two-way speakers is evident because the woofer is used only up to 700 Hz, where it is still nondirectional. Wide midrange directivity takes over and blends with the tweeter's output.

Figure 7 is the vertical off-axis plot—from below, to the front, to directly above the speaker. Since different angles in this plane produce different distances from the drivers to the microphone (or ears), I expect to see angles where the drivers interfere at the crossover frequencies. There are virtually none near either 700 Hz or 6.5 kHz. This excellent performance is due to good choice of frequencies and the high slope of the crossovers.

Figures 8, 9, and 10 indicate nonlinearity by plotting harmonics generated in the loudspeaker when reproducing a pure tone. There are 3-D plots, with the front-to-back axis indicating power input from 0.1 to 125 watts; the second through the fifth harmonics are shown. The 3.8D is excellent at the test frequencies of 41.2, 110, and 440 Hz (the musical notes E_1, A_2, and A_4, respectively). With 440-Hz input (Fig. 10), the fourth harmonic is almost entirely produced by the midrange, which receives some 440 Hz despite the crossover's 700-Hz high-pass filter.

Figure 11 plots the modulation of 440 Hz by 41.2 Hz when both are fed to the speaker in equal amounts. This is essentially another linearity test at 41.2 Hz, but other factors—such as frequency modulation—are introduced. The 3.8D does well, but it has its power limits. Distortion climbs rapidly above 20 watts and is an annoying 10% at 70 watts.

Power linearity is a full-range linearity test. Starting with 1.0 watt, power is increased in steps to 256 watts. Figure 12 shows the power input and the frequency where the acoustic output fails to track the input increase by 1 dB. Amplifier power greater than this amount will be of little benefit.

A speaker with this smooth response does not grab you with specific attributes. It may take time to fully appreciate the 3.8D.

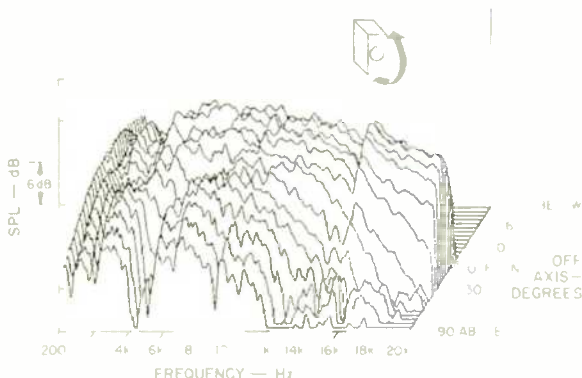

Fig. 7—Vertical off-axis plots taken from below, up the front, and to the top. Note relative absence of interference effects at the crossover frequencies of 700 Hz and 6.5 kHz.

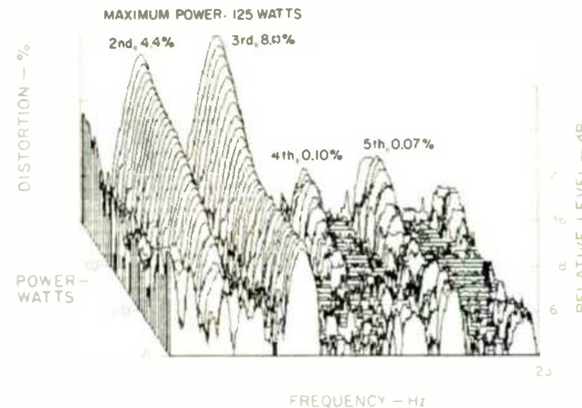

Fig. 8—Harmonic distortion products for the tone E_1 (41.2 Hz).

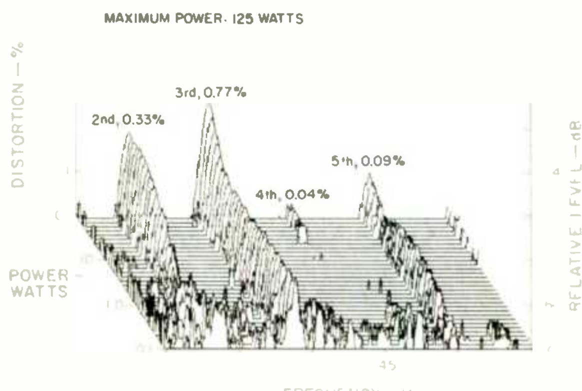

Fig. 9—Harmonic distortion products for the tone A_2 (110 Hz).

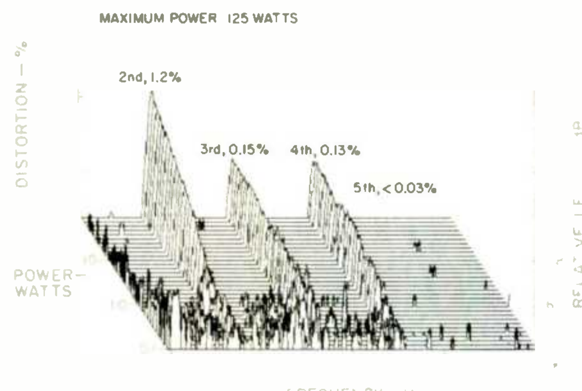

Fig. 10—Harmonic distortion products for the tone A_4 (440 Hz).

Although the 256-watt maximum applied is handled over much of the 3.8D's range, the manufacturer's 160-watt recommendation is certainly reasonable.

The energy versus time response is shown in Fig. 13. In this test, energy is at its maximum in the upper part of the test range of 20 Hz to 20 kHz. The plot shows the strong tweeter output and shows that later arrivals, from diffraction and reflection, are more than 24 dB down.

Use and Listening Tests

The most exciting part of a loudspeaker review is the listening evaluation, but preceding this are the mundane tasks of unpacking the speakers and reading the manual. I mention this because the 3.8Ds had the most elaborate and protective packing I have seen for a loudspeaker. A beautifully made wooden frame snaps onto the front of the speaker for protection, followed by plastic wrapping, ⅛-inch sheets of plywood for all six sides, a corrugated box, Styrofoam isolators, and finally, the outer corrugated shipping box. (Grilles are shipped separately.) I was impressed and ready for great sound.

The manual recommends placing the 3.8Ds on short stands, away from walls. Fuselier provides simple room-ratio formulas to suggest a starting position. Obviously, the manufacturer expects that the user is an audiophile willing to meet the demands of best sound reproduction. I found positioning to be relatively noncritical, meaning that suboptimal positions did not degrade the sound greatly. Best positioning proved to be 3 feet out from an 18-foot wall and 4 feet away from the side walls. The Fuselier 3.8Ds were placed on stands and rotated inward to minimize the side-wall reflections.

I recommend the Fuselier 3.8D as a refined loudspeaker for the musically mature audio enthusiast.

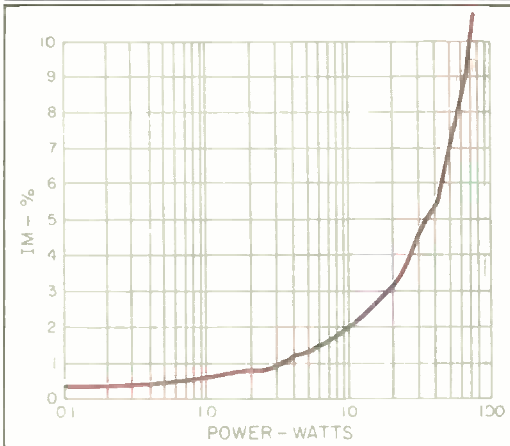

Fig. 11—IM distortion on 440 Hz (A_4) produced by 41.2 Hz (E_1) when mixed in one-to-one proportion.

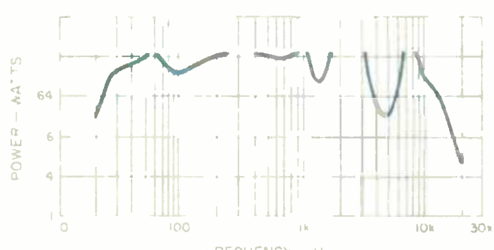

Fig. 12—Power linearity (input power handling vs. frequency for 1-dB compression of the output).

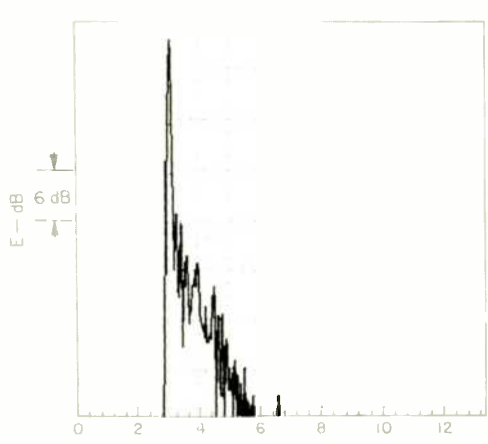

Fig. 13—Energy vs. time.

I listened to the 3.8Ds using a variety of familiar recordings, mainly jazz and classical. (I avoid pop and synthesizer music when evaluating naturalness and fidelity.) Spectral uniformity of the 3.8Ds was outstanding, with low coloration and good overall balance. Octave-to-octave balance was only slightly upset by excessive high-frequency content and lack of the lower part of the octave from 16 to 32 Hz. Bass power, extension, and freedom from distortion were up to the highest standards for listening to natural music at reasonable levels, the one minor exception being the lowest organ pedal tones. In other words, the bass is better than that of most add-on subwoofers.

The measured high-frequency rise was a surprise to me when I first saw it, because excessive highs had bothered me only slightly in the listening evaluation. My conclusion, therefore, is that the extreme smoothness of the response makes this rise acceptable. Loudspeakers with a rougher response and boosted highs are always irritating to me.

Proper spatial rendition requires that the speakers "disappear" as perceived sources of sound. The listener should be able to visualize a wide, but not too wide, soundstage containing stable instrument images and a sense of correct ambience. The system should correctly render *differences* in perceived depth of images in the soundstage, not just a limited perspective that is always close or always distant. Success in creating these perceptions depends heavily on the recording, the listening room, and speaker/listener placement. I use proven good recordings, an excellent room, and optimal positioning to get the most from the loudspeaker systems.

The 3.8Ds excel again in spatial rendition: They are as good as, and perhaps better than, any system I have had in my listening room. They were particularly impressive playing the Arrigo Boito Prologue to *Mefistofele* (Telarc CD-80109-2). The orchestral, choral, and organ sections of this thickly textured piece were separated and arrayed evenly across, and deeply into, the soundstage. Ambience reproduction was perhaps just short of the best I have heard. Dipole speakers and simple ambience extraction schemes have given me the best results so far.

Loudspeakers with a "presence peak" could have more definition and clarity than the 3.8Ds, but this would obviously be wrong. Also, elevated upper bass could give more "punch." Again, wrong. A loudspeaker with a smooth response, like the 3.8D, does not grab you with one of these focused attributes, and so it may take time to fully appreciate this speaker. If you want punch or presence from these systems, use an equalizer, or better yet, just turn up the volume. The 3.8Ds will play quite loudly with no sign of distortion.

John Fuselier told me that the sample units sent to *Audio* for review do not reflect a slight reduction in high-frequency energy that has been applied in production for some months now. In any case, elevated highs might be just right for more distant listening in an absorptive room. My treble nit-picking—dare I admit that I have a treble control and occasionally use it?—is a very minor issue. I enjoyed my music through these speakers and recommend the Fuselier 3.8D as a refined loudspeaker for the musically mature audio enthusiast.

David L. Clark

The Next Generation

The DQ-20

Imagine a speaker system that delivers transparent imaging, accurate dimensionality in width, height and depth, combined with harmonic integrity and dynamic power.

The Dahlquist DQ-10, the original Phased Array dynamic speaker system, became a legend in its own time. This seminal design employed many of the concepts which are still at the leading edge of loudspeaker design. The importance of low diffraction distortion and correct inter-driver time delay were certainly popularized by the DQ-10.

The DQ-10 has been replaced by the DQ-20. Three extraordinary drivers have been combined with advanced enclosure technology to cover a wider range with greater efficiency than was possible with the DQ-10. The same attention to diffraction control and time delay distortion allows the DQ-20 to provide the expanded open window on the soundstage for which Dahlquist is famous.

Clearly superior, the DQ-20 stands as the ultimate stereo vehicle that will transport you into the realm of pure sound. This incredible achievement lets you surround yourself with the captivating reality of brilliant musical performances and listening pleasure.

It is an experience you will never forget.

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6

SONY CDP-707ESD COMPACT DISC PLAYER

Manufacturer's Specifications

Frequency Response: 2 Hz to 20 kHz, $\pm$0.3 dB.

Signal-to-Noise Ratio: 115 dB (EIAJ).

Dynamic Range: 100 dB (EIAJ).

THD: 0.0015% (EIAJ).

Separation : 110 dB (EIAJ).

Wow and Flutter: 0.001% wtd. peak (EIAJ).

Number of Programmable Selections: 20 (see text).

Number of Discs Unit Can "Recognize": 226.

Number of Customized Index Points per Disc: 6 (see text).

Output Level: Fixed, 2.0 V rms; variable, 0 to 2.0 V rms.

Digital Output: Coaxial, 0.5 V peak to peak; optical, per EIAJ Standard.

Headphone Output Level: 28 mV.

Power Requirements: 120 V a.c., 60 Hz, 18 watts.

Dimensions: $18\frac{1}{2}$ in. W $\times$ $4\frac{15}{16}$ in. H $\times$ $14\frac{3}{4}$ in. D (47 cm $\times$ 12.5 cm $\times$ 37.5 cm).

Weight: 39.6 lbs. (18 kg).

Price: $1,800.

Company Address: Sony Dr., Park Ridge, N.J. 07656.

(Originally published April 1988)

If you have been wondering just how much more a manufacturer could do to extract every last bit of recorded information from a Compact Disc without introducing extraneous noise or distortion, I commend to you Sony's latest state-of-the-art CD player, the CDP-707ESD. Frankly, I am still a bit dazed after measuring and listening to this unit. I thought that the industry had gone about as far as it could with some of the latest high-end CD players that have appeared recently. I was wrong. The CDP-707ESD beats anything I've ever measured or heard—and not by a small margin, either. I am just thankful that it arrived *after* I acquired my Audio Precision System One test gear; otherwise, I might never have been able to measure the residual noise and distortion levels that this incredible player exhibits.

Before I tell you about the measurements and the sonic excellence of this unit, I want to discuss some of its more attractive convenience features. Several of these are almost beyond belief, but trust me, they all work as described.

Perhaps the most outstanding of these are the "Disc Memo," "Custom Index," and "Program Bank" features. "Disc Memo" allows you to compose a message of up to 10 letters, numbers, or symbols and have the message memorized with respect to a specific disc. The stored message is then shown in the front-panel display whenever you load that particular disc. If you wish, the message can be displayed during play as well. "Custom Index" allows the player to memorize up to six index points on each disc as a way of marking your favorite musical passages. These personal index points are in addition to any specific index points that the record company may have encoded onto a given CD, and they can be applied even to discs that have not been so encoded by the manufacturer. What a boon for those of us who own a lot of early CDs, many of which are without index points. The "Program Bank" allows you to program favorite selections from a given disc into the CDP-707ESD's memory, in any order you choose. Once you do that, any time you load a disc for which a program has been "banked," the unit will automatically play only those selections you have stored. Any of these features can be applied to as many as 226 different discs. "Disc Memo" and "Custom Index" or "Disc Memo" and "Program Bank" may both be filed for any given disc, but you cannot use "Custom Index" and "Program Bank" simultaneously for a disc.

If all of this sounds as though the age of the read/write or erasable CD has arrived, rest assured that it has not. You see, every CD that's ever been made has its own identifying code. When you use any of the three features I've just described in the Sony CDP-707ESD, the player reads the unique identifying code associated with the disc and stores that code along with your programming or memo instructions. Then, the next time that disc is loaded, the player's microprocessor matches the code to the programs that have been stored for that code.

Of course, the more familiar operating features found on earlier Sony (and other) CD players are here too. Up to 20 tracks (or tracks plus index points) can be programmed in any order for immediate playback. As a further enhancement, you can also perform a "Program Edit." That is, you can program a sequence of selections with a built-in pause point. For example, you might create a program 50 minutes long with a built-in pause at 25 minutes. During playback, the CD will automatically pause at the end of the first part (25 minutes) and will resume, to play the second part of your program, when you press the "Pause" button.

"Disc Memo" allows you to assign a message of up to 10 characters to each CD; every time a disc is loaded, its "memo" is displayed on the player's front panel.

The popular "Shuffle Play" (random play) feature introduced by Sony more than three years ago has been retained, and it too is augmented. Now you can instruct the player to "Shuffle Play" only those tracks or selections that you want. This feature, called "Delete Shuffle," is aptly named: After requesting "Shuffle Play" mode, you designate the tracks that you *don't* want included by its use. Incidentally, this unit will play the 3-inch CDs too.

Fast audible search is available, and access time to any point (track or index) on a disc remains under 1 S, as in earlier top-of-the-line Sony CD players.

Circuit Refinements and Construction

For all of these useful and impressive convenience features, of even greater significance, in my opinion, are the circuit improvements and structural refinements that have been incorporated in the CDP-707ESD.

This player is the first that I know of to employ eight-times oversampling. Sampling, in a digital audio system, is the process of looking at incoming waveforms as a series of discrete events in time. The speed of this process is the sampling rate. In a digital filter, the sampling rate is the rate at which the filter reads and calculates binary values, outputting them as a data stream to the D/A converter or converters.

In general, as the sampling rate increases, high-frequency performance is improved, output waveforms become more linear, and phase shift and losses due to steep analog filtering at the output of the system are minimized. The use of digital filtering and oversampling doesn't provide any more *real* data from a Compact Disc, as some have suggested. Rather, the filter creates its own additional data, producing a given number of "calculated" output levels for every "real" value read from the CD (which carries data

This Sony player employs true 18-bit D/A conversion, and it is the first one I know of with eight-times oversampling.

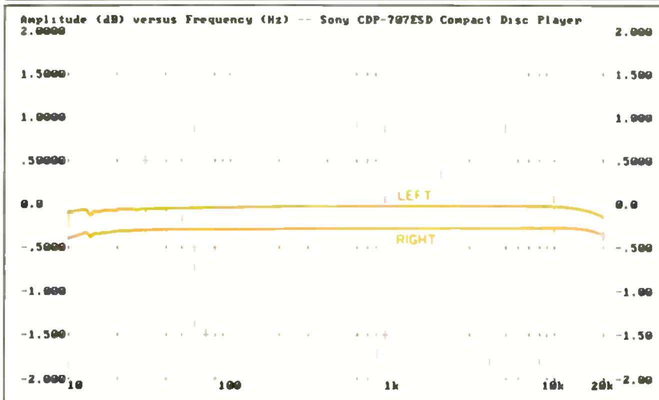


Fig. 1—Frequency response of left and right channels; curves have been separated for clarity.

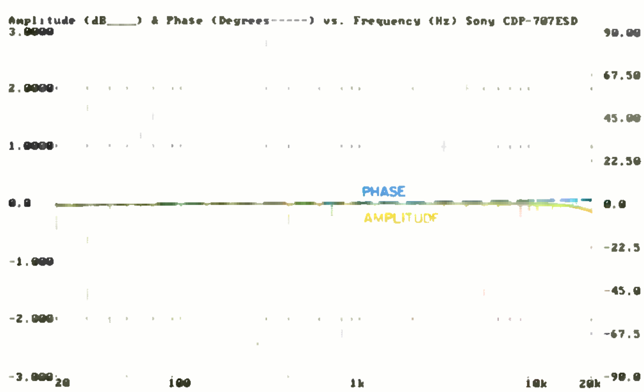


Fig. 2—Comparison between amplitude response of one channel (solid curve) and relative phase response (dashed curve) of opposite channel. Phase, in degrees, can be read from right-hand scale.

recorded at the standard sampling rate of 44.1 kHz). In doing so, it makes for a smoother, more linear waveform while at the same time shifting unwanted modulation noise farther out in frequency, where it can be more easily filtered out by simpler analog filters.

The digital filter chip that Sony has developed (identified as a CXD-1144) employs 18-bit technology to match the 18-bit D/A converters used in this model (about which more in a moment). Oversampling at eight times the 44.1-kHz CD sampling rate results in a very dense data stream. This increased density improves linearity even beyond that achieved by four-times oversampling.

By employing true 18-bit D/A conversion, the CDP-707ESD takes advantage of the increased data density available from the digital filter. Both the D/A converter and the digital filter use full 18-bit processing, so there is no need to employ bit register shifts such as those used in the quasi-18-bit systems. Eighteen-bit linear conversion of the 16-bit signals contained in a CD provides 12 dB more signal-to-noise headroom, since the lowest two bits are never actually required to describe a signal amplitude. The number of possible quantization values in a 16-bit system is 65,536; adding two bits (as in 18-bit D/A conversion) increases that number to 2^{18}, or 262,144. Clearly, having this many incremental values available reduces distortion at lower levels, because quantizing values are available at the bottom of the "bit" scale, so to speak.

Signal-to-noise capabilities are calculated by allowing just over 6 dB of S/N for each bit. Sixteen-bit systems, therefore, can have a maximum wide-band signal-to-noise ratio of 98.1 dB, while 18-bit linear converters have a maximum S/N capability of 110.1 dB.

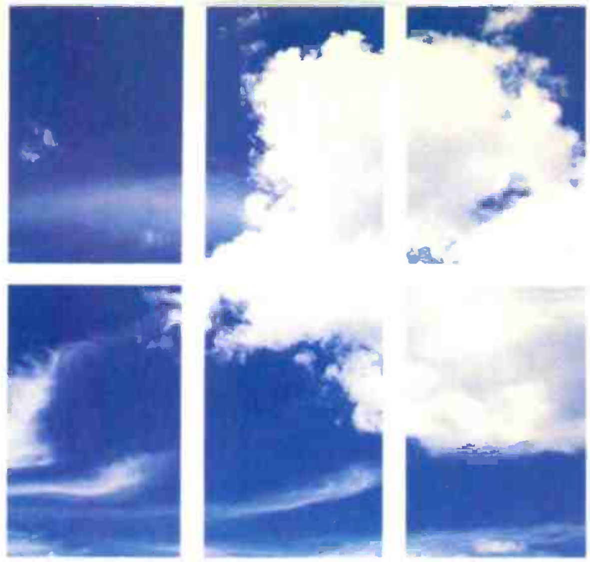
It should also be noted that there are 14 power-supply subregulation stages within the CDP-707ESD. The digital output terminal has been configured in accordance with newly adopted Sony/Philips Standards for CD graphics as well as CD audio. The optical output conforms to the recently agreed-upon EIAJ Standard for optical digital interfacing.

The base of this player is built from multiple layers of both copper and Sony's "G" material (plastic resin and calcium carbonate, reinforced with glass fiber to resist vibration). These layers are then bonded to a steel plate for added rigidity. The base is acoustically, magnetically, and electrically inert. Dual-wall construction is also used; a second inner side wall of copper plating completely encloses the player, providing r.f. and magnetic shielding. Plastic-sealed wire-harness openings prevent the transfer of vibration from one stage to the next. As in previous models, the linear motor that drives the laser pickup assembly has been mounted in a double suspension system; part of the base or mounting surface is made of a ceramic/resin compound to reduce vibration and resonance caused by the laser assembly's fast motion.

The dual transformers and the motor assembly are mounted in a separate chassis compartment to isolate the power supply and the drawer mechanism from digital and analog circuit boards. As in earlier Sony units, circuit refinements such as single "master-clock" architecture and an advanced error detection and correction system have been employed. Finally, the analog filters used in the output stage are of a linear-phase design and employ only a three-pole architecture.

Control Layout

"Power" and "Timer" switches are at the far left of the front panel, beneath the disc drawer. Major pushbuttons running across the lower portion of the front panel include those for drawer "Open/Close," play, pause, and stop, plus "AMS" (Automatic Music Search) buttons for moving ahead

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Overall S/N when playing a quiet test-disc track was the highest I've ever seen: 115 dB unweighted and 118 dB A-weighted.

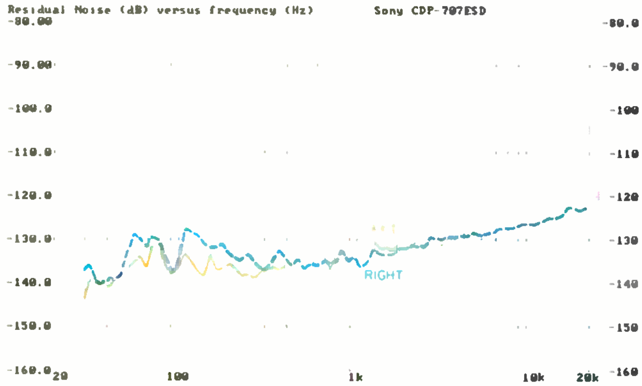


Fig. 3—Residual noise vs. frequency for "quiet" track of CD-1 test disc. Note low noise level at 60-Hz power-line frequency and its multiples (see text).

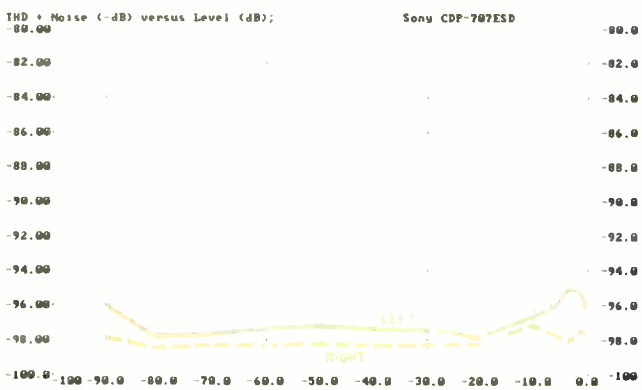


Fig. 4—Quantization noise and distortion vs. signal level.

or back to the next or previous track. Fast-forward and fast-reverse search buttons come next, followed by an "Erase" button. This is used to erase information stored in programming or in any of the "files" created for your discs when using the "Disc Memo," "Custom Index," "Program Bank," "Program Edit," or "Delete Shuffle" features.

Above the main transport buttons is a large display area that includes a "calendar" indicating the status of up to 20 tracks, the alphanumeric "Disc Memo" message, readouts for various modes of CD playing time (track, total, or remaining time), a readout for programming data, and much more. To the right of the display are 20 small, numbered keys used both in accessing a particular track and in programming. Still farther to the right are the headphone jack, a level control for the variable line outputs, and buttons for selecting the "Continue/Single," "Shuffle," and "Program" play modes.

A row of small keys controlling secondary functions is arranged between the display area and the primary controls. These secondary buttons include "Display" (for recalling "Disc Memo" notations and the like), "Space/Pause," "Repeat," forward and reverse "Index," ">20" (used to access track numbers higher than 20), "Check," "Clear," "File," and "Index Mode."

The rear panel is equipped with fixed and variable line-level analog output jacks, a coaxial digital output jack, an optical digital output jack, and a switch that selects either analog or digital outputs.

The player comes with an infrared remote control which duplicates nearly all of the control and programming functions found on the front panel. The remote even has a pair of pushbuttons for controlling output level if the player is connected to an amplifier via the variable output jacks.

Measurements

Just as I was getting ready to measure this player with the aid of my newly acquired Audio Precision System One test equipment, the good people at Audio Precision—almost as if on cue—sent an application note on CD testing along with a floppy diskette full of newly devised tests and procedures. These new procedures involve the CD-1 CD test disc, which conforms to the proposed EIA Measurement Standard that I have been working on with other members of the audio industry for the past couple of years. This disc, produced by CBS Records, has some important test tracks providing information about CD player performance that no other test discs yield. Needless to say, the ability to run these tests on Audio Precision software meant that I didn't have to create test panels on my computer screen to go with the Audio Precision System One hardware.

Figure 1 shows the CDP-707ESD's frequency response from 10 Hz to 20 kHz. Even with an expanded scale of 0.5 dB per division, the curves follow an almost perfectly straight line over most of the audio range. Deviation from perfect flatness was −0.15 dB at 20 kHz. (Curves of left and right response have been deliberately separated for clarity; actual output levels for the two channels were within 0.1 dB of each other.)

In Fig. 2, I plotted the amplitude response of one channel (solid curve) versus the relative *phase* of the opposite channel (dashed curve). As you can see, there was *no* detectable phase error between channels, even at 20 kHz!

Overall signal-to-noise ratio of the CDP-707ESD, when playing a "quiet" track of the CD-1 test disc, was the highest I have ever measured for any CD player: 115 dB unweighted and 118 dB A-weighted! It should be understood that this test produces no information concerning the *digital* portions of the system, since there is no data on the "quiet" track to exercise the D/A converters.

Amazingly, instead of increasing at higher frequencies, THD + N actually decreased as 20 kHz was approached.

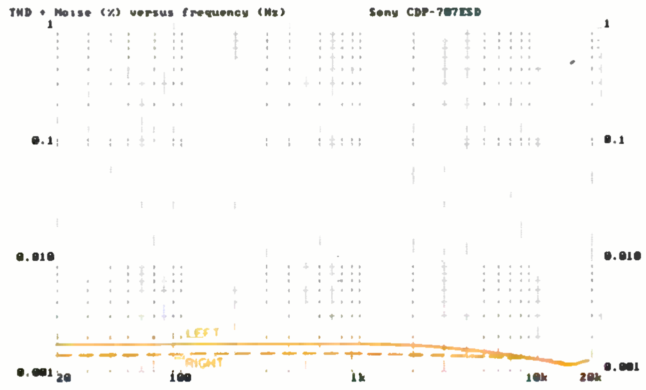


Fig. 5—Quantization noise and distortion vs. frequency. Note decrease at high frequencies (see text).

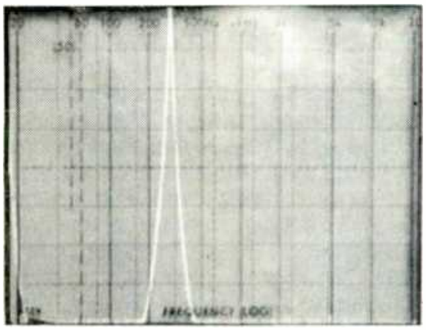


Fig. 6—Spectrum analysis of 20-kHz signal; note absence of 24.1-kHz "beat" tone seen on many other players. (Vertical scale: 10 dB/div.)

A spectral analysis of the residual noise will show whether a CD player's power supply is adequately filtered and shielded. On one or two other players that I have subjected to this test, I could see rather large peaks of noise at 60 and 120 Hz and, in some cases, at 180 Hz, all attributable to the players' power supplies. Notice that in Fig. 3, the CDP-707ESD's residual noise at these power-supply related frequencies is −130 dB or lower in every case but one, and just a bit above −130 dB in that one case.

Quantization noise and distortion, on the other hand, are due to errors (binary round-off or approximation) that occur during the digitization process as the player converts a recovered analog signal's amplitude to the nearest available number. If all other forms of noise and distortion were negligible, quantization noise and distortion would set a "floor" (−98.1 dB for the CD's 16-bit linear system) that would be constant in absolute magnitude for all signal levels up to 0 dB. The graph of Fig. 4 expresses quantization noise and distortion level in dB (vertical scale) against signal amplitude from maximum (0 dB at the right) down to −90 dB. At all but the very highest levels and the very lowest level, the quantization noise and distortion come remarkably close to the theoretical 98.1 dB. The slight rise at the high end of the plot for the left channel (solid curve) may well be due to a minute amount of distortion in the analog amplifier stage. Bear in mind, however, that even −95 dB corresponds to 0.00178%!

An alternative method of presenting THD + N versus output amplitude is to express it as a *percentage* of the available signal level. Of course, when presented in this way, THD + N rises linearly with decreasing signal levels. Fixed, spot readings of this player's THD + N at 0-dB recorded level were 0.00135% on one channel and 0.0014% on the other—again, the lowest results I have ever obtained for any CD player.

Perhaps even more amazing is the fact that this THD + N level, rather than increasing at higher frequencies because of nonharmonically related out-of-band "beats," actually decreased as the test frequencies approached 20 kHz, as shown in Fig. 5. To confirm this phenomenon, I applied a 20-kHz test signal to a spectrum analyzer, as I have been doing now with most CD players I test, to generate the 'scope photo of Fig. 6. The sweep is, as usual, linear from 0 Hz to 50 kHz, but there is absolutely no evidence of the 24.1-kHz beat so commonly seen from most players. I even stepped up the gain of the analyzer to just below its own clipping level in order to display a full 80 dB of dynamic range on the 'scope. (I usually settle for 70 dB, in order to leave a safety margin for the analyzer's circuits.) Even then, there were absolutely no spurious products—only the desired signal at 20 kHz, where it belonged. If I had any doubts about the benefits of eight-times oversampling or of 18-bit D/A conversion and digital filtering, Figs. 5 and 6 dispelled those doubts completely!

Figure 7 is a plot of separation between channels; the results from left to right and from right to left were so close to each other that I did not bother to identify which curve is which. Suffice it to say that at 1 kHz, separation in either direction was greater than 125 dB, and that at 20 kHz, where separation often decreases to 70 dB or less, it still measured well over 100 dB!

Next, I investigated an important (and excellent) characteristic of the CDP-707ESD, its linearity. When reproducing undithered test signals, the player showed virtually no deviation from perfect linearity from 0 dB (maximum level) down to about −80 dB. In other words, output levels were almost perfectly proportional to the signal levels recorded digitally on the test disc. There was a very slight departure from linearity at −90 dB, but this was very hard to see in a

When I got the unit to mistrack, by hitting it rather hard, there was no skipping around but just a brief moment of muting.

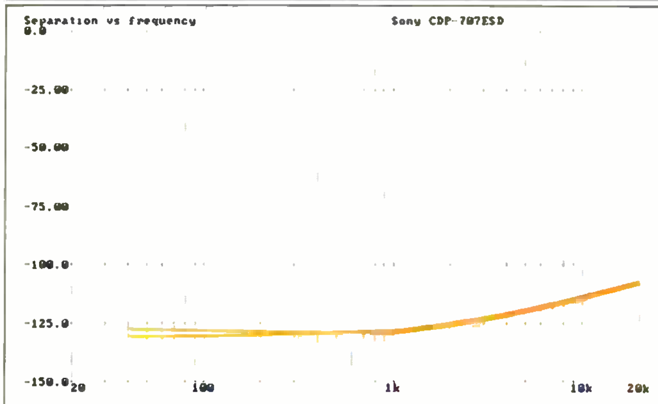


Fig. 7—Interchannel separation. Note the unusually close match between the two curves (see text).

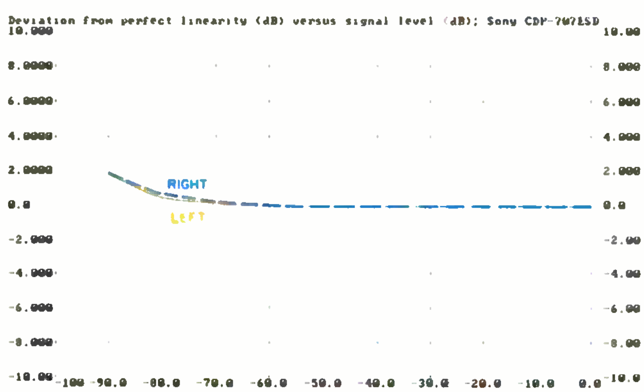


Fig. 8—Deviation from perfect linearity for undithered 1-kHz signal at levels from −90 to 0 dB.

conventional linearity graph (output versus input amplitude). This is where one of the "procedure" files supplied with my new CD test software comes in. In a series of computations and corrections, the file translates the measured output-versus-input linearity data into a graph that shows *deviation* from flatness with decreasing level. Here, in Fig. 8, it is much easier to see that at −90 dB, linearity was off by a mere 2.0 dB.

The CD-1 test disc contains low-level signals (from −70 to −100 dB) to which dither noise has been added. Although dither reduces the ultimate signal-to-noise ratio of a recording, it also reduces distortion. To put it another way, it extends linear operation below the undithered theoretical limit. Signals whose peak amplitude is less than one-half the least significant bit's value could never be recorded in a nondithered system. With dither, signals at such low amplitudes are still converted, since the dither acts to ensure that the lowest significant bit is always exercised. If this seems hard to fathom, think of dither in the frequency domain as spreading the quantization noise across the spectrum, rather than having all the energy concentrated at harmonics of the desired signal frequency.

In any event, Fig. 9 is a plot similar to that of Fig. 8, but using signals which are dithered and which extend only from −70 to −100 dB. Again, the deviation from perfect linearity is only about 2 dB (a bit more in one channel, but a bit less in the other) *at −100 dB!*

Two methods for measuring a CD player's dynamic range have been proposed. The first, offered by the Electronics Industries Association of Japan, involves measuring the THD + N of a −60 dB signal, expressing it in dB, and adding the negative number of dB to −60 dB. Spot readings of my test equipment yielded an EIAJ dynamic range of approximately 98 dB for the CDP-707ESD.

A second method, proposed by the Electronics Industries Association, utilizes a special dithered signal on the CD-1 test disc that "fades" from −60 to −120 dB (into the noise). Dynamic range is then considered to be the difference between 0 dB (maximum level) and the point during the fade where the signal level is 3 dB higher than the lowest point it ever reaches. In the past, I have tried to measure this point by eyeballing an a.c. voltmeter, but I've found that, at best results have been none too accurate. Thanks to another "procedure" file created by Audio Precision, the task became quite simple. The system performed a complex series of tests, computations, translations from time frame to level, and the like. During the first stage of this multiple-test procedure, I was able to easily pick out the reading that was 3 dB above the minimum reading; it was −111 dB, which therefore is the EIA dynamic range of this CD player. The final results of this procedure are plotted in Fig. 10, a graph of departure from perfect linearity down to −120 dB.

Despite the greatly increased sensitivity of my new test equipment over the gear I had been using, the wow-and-flutter test, conducted as a function of time for a period of 25 S, yielded a line along the "0" baseline of the grid on which it was plotted. As they so often say in the published specs, wow and flutter was "below measurable limits"—even though those limits have now been extended downward by a couple of orders of magnitude.

SMPTE IM *was* measurable. I read 0.006% for one channel and 0.005% for the other. CCIF twin-tone distortion, using 11- and 12-kHz test signals from the CD-1 test disc, was 0.00123% on either channel. Clock accuracy—the accuracy of the player's 44.1-kHz master clock—is another parameter I am now able to test. Any gross errors in clock accuracy would result in a deviation from correct musical pitch when playing CDs. No such worry with the CDP-707ESD: Its clock was off by only 0.0175%. To put it in

I was able to figure out just about all of this player's features without trouble, even lacking an English owner's manual.

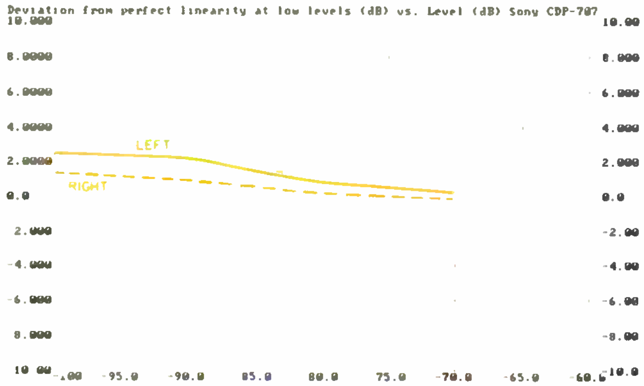


Fig. 9—Deviation from perfect linearity for dithered 1-kHz signal at levels from −100 to −70 dB.

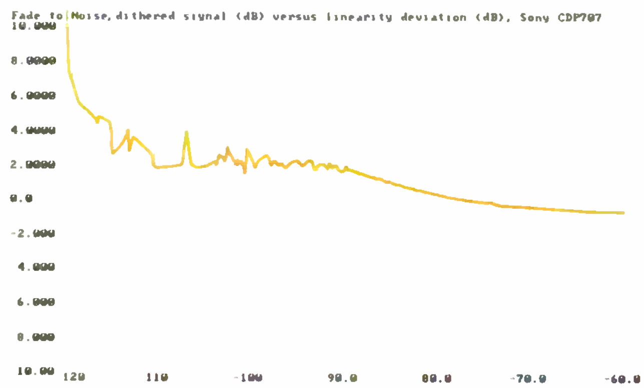


Fig. 10—Linearity deviation for EIA "fade-to-noise" dynamic-range test, using dithered signal.

perhaps more meaningful terms for those of you who study music, a 440-Hz tone (A above middle C) reproduced on this player will come out as 440.07 Hz.

Figures 11 and 12 show how the player reproduced a 1-kHz square wave and a unit pulse, respectively. The slight ripple on the horizontal portions of the square wave are due to the CD system's bandwidth limitation (cutoff at 20 kHz) and not to any other cause. The unit pulse merely confirms that this player does not invert polarity.

Use and Listening Tests

The Sony CDP-707ESD is so well constructed, and its laser pickup assembly so well isolated from the rest of the unit, that even rather severe pounding on its top surface failed to cause any mistracking. When I finally got the unit to mistrack, by hitting its side panels rather hard, it did not skip all over the disc, as other players have done. Instead, there was only a brief moment of muting and then a resumption of play at almost precisely the point where the music had been interrupted.

The "Disc Memo" feature is intriguing. I don't know what 10-letter messages people will assign to their discs, but if nothing else, the feature is fun to use and experiment with. The "Custom Index" function, on the other hand, I consider to be extremely useful. I own many discs to which I would like to assign my own index points, in order to more easily find musical passages that I enjoy listening to or that I use in testing audio equipment performance. With the "Custom Index" feature, I could identify these passages easily and recall them accurately. It is a credit to this unit's designers that although the English-language owner's manual was unavailable when I tested the CDP-707ESD (such manuals are, of course, available now), I was able to figure out just about all of its regular functions—and even its unique special features—without too much trouble.

When you get beyond all the clever features, a top-of-the-line CD player is expected to reproduce music as accurately and as faithfully as the CD medium will permit. To my way

Even if you can't afford one, you should listen to the CDP-707ESD. It is a shining example of what CD technclogy's all about.

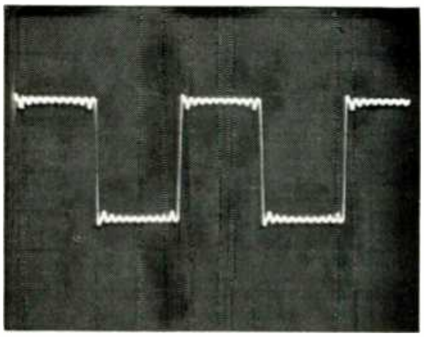


Fig. 11—Reproduction of 1-kHz square wave.

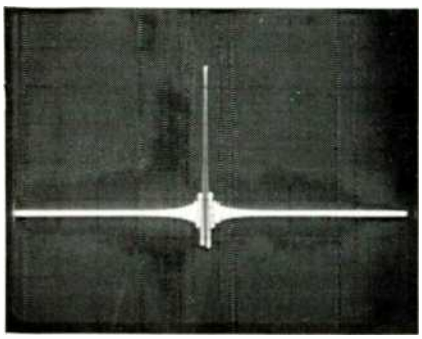
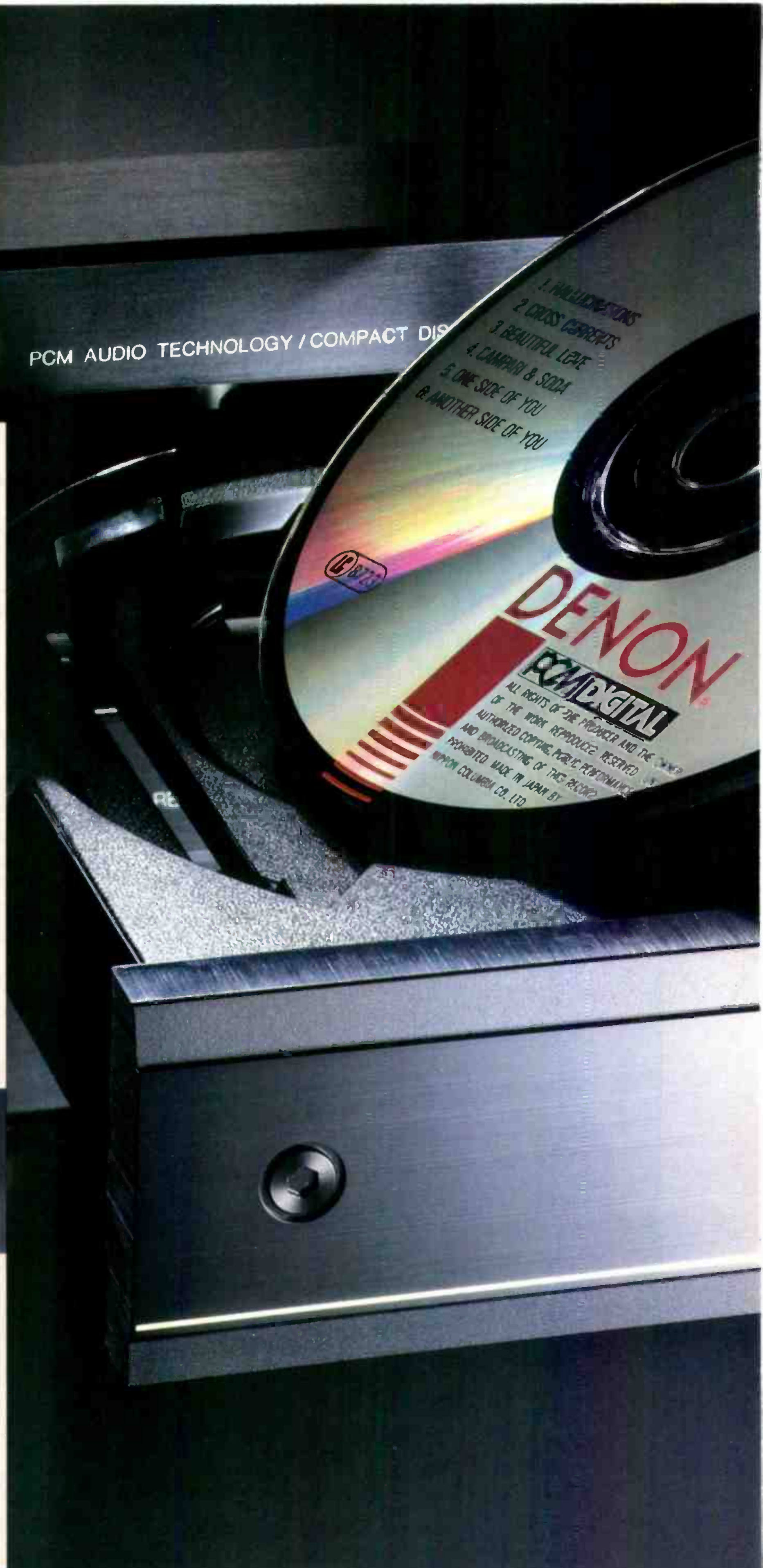


Fig. 12—Single-pulse test.

of thinking, the CDP-707ESD does just that. I will not list all of the discs I auditioned on this player; to do so would make an already long report even longer. Three that stand out, though, are a Telarc disc (CD-80142) of Mendelssohn's "Quartet in A Minor" and his "Octet in E Flat Major," a Delos disc (D/CD 3062) which contains Haydn's Symphony No. 21 in A Major as well as his Cello Concerto No. 1 in C Major, and another Telarc disc (CD-80155) of Brahms' Symphony No. 4 in E Minor as well as his "Academic Festival Overture." There's enough variety on these three discs to put any audio system through its paces. I particularly like to play music of small instrumental groups when judging a CD player or, for that matter, any audio component. The clarity and warmth of the cello on that Delos disc has never been reproduced more faithfully than on this Sony player. Even if you can't afford a CDP-707ESD, you owe it to yourself to listen to how it performs and to try out its unique features.

If nothing else, the CDP-707ESD should serve us all as a reference against which to judge the new crop of players that will appear in the coming months. Perhaps they too will employ true 18-bit D/A converters (some already do), eight-times oversampling with 18-bit digital filters, and all the other circuit innovations found in this unit. But until they arrive, the Sony CDP-707ESD stands alone as a shining example of what Compact Disc technology is all about.

Leonard Feldman



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7

ONKYO DX-G10 COMPACT DISC PLAYER

Manufacturer's Specifications

Frequency Response: 2 Hz to 20 kHz (no tolerance given).

Sampling Frequency: 352.8 kHz (eight-times oversampling).

THD: 0.0015% at 1 kHz.

Dynamic Range: 103 dB.

S/N: 110 dB.

Channel Separation: 103 dB at 1 kHz.

Number of Programmable Selections: 16.

Output Level: 2.0 V rms.

Power Requirements: 120 V a.c., 60 Hz, 24 watts.

Dimensions: $18\frac{3}{4}$ in. W × $5\frac{9}{16}$ in. H × $16\frac{13}{16}$ in. D (47.7 cm × 14.2 cm × 42.7 cm).

Weight: $59\frac{1}{2}$ lbs. (27 kg).

Price: $2,500.

Company Address: 200 Williams Dr., Ramsey, N.J. 07446.

(Originally published March 1989)

When the double-cartoned Onkyo DX-G10 first arrived at my laboratory, I was sure it was either a power amplifier or a high-powered integrated amp. After all, who would have suspected that a CD player would weigh nearly 60 pounds? But after carefully unpacking this heavyweight, it was soon obvious that this was, indeed, a Compact Disc player. It gave me the impression, even before I tested it or listened to it, that Onkyo's management must have said to its engineers, "Forget about cost! Just design the finest, most rugged, and technically sophisticated CD player you can. Let us worry about whether or not we can sell any at a $2,000+ price." In large measure, those engineers have succeeded.

Consider, first, this unit's extremely heavy weight. The DX-G10 is housed in an extremely rigid, graphite-reinforced steel chassis. The laser pickup mechanism and all modular circuit blocks are firmly anchored to this foundation. Onkyo maintains that this arrangement not only isolates the circuitry and mechanisms from any outside or external vibrations, but that it absorbs internally generated vibrations from the disc-drive motor, power transformers, and other sources.

The laser pickup assembly uses a total of four motors, each optimized for its specific task. The most critical of these, according to Onkyo, is the pickup drive motor. Here, the DX-G10 uses a linear motor capable of high-speed access.

As for circuit innovations, this is not the first time Onkyo has used a true linear 18-bit D/A conversion system, but in this unit, they have combined it with some pretty fancy optical circuitry which they call "Opto-Drive." It serves as the signal-current source for the converters, substituting an LED and phototransistor for the more conventional zener-diode current source. The phototransistor's output, says Onkyo, is far more stable than a zener's, and the optical coupling reduces the amount of interference reaching the D/A converters. To eliminate phase differences between stereo channels, two ladder-network D/A converters are used.

Another feature, which Onkyo calls "Opto-Coupling," involves electrical isolation of the digital and analog circuitry

through the use of fiber optics. A total of five discrete optical fibers form a sort of "light bridge" or data link between the digital and analog sections of the player. The analog circuit blocks are fully shielded against any high-frequency interference from digital pulses. Just how much all of these internal refinements contribute to the ultimate sound quality of the player is something each listener will have to determine for himself, but I'll have more to say about it later on.

The unit comes with a full-function remote control. In addition to fixed and variable analog outputs, the DX-G10 has both optical and coaxial digital outputs, and a now-standard optical cable is supplied to hook up the digital output directly to an optical digital input on a stand-alone D/A converter or on one of those new amps which have built-in converters. (In view of the details lavished upon the Onkyo's built-in D/A conversion system, I wonder why anyone would go to that additional expense.) A "Shuttle Search" knob is used for fast-forward or reverse seeking of a specific moment of music. Various repeat-play modes are available, including A-to-B. Index points within a track may be directly accessed, and programmed play of up to 16 selections can be memorized by the player's microprocessor. You can even instruct the DX-G10 to begin play from a specific point (in minutes and seconds) within a given track by simply punching in the track number and the time.

Control Layout

Despite its rather massive appearance, the front panel of the DX-G10 is not terribly cluttered. That's because only the most used controls and pushbuttons are visible when the full-width drop-down door, along the lower edge of the panel, is in its closed position. A power switch and indicator are at the extreme left, adjacent to the disc tray. To the right of the tray is the display area; it shows track and index numbers, various time readouts, and other status indica-

Onkyo must have asked its engineers for the finest, most rugged, sophisticated CD player they could make, regardless of cost.

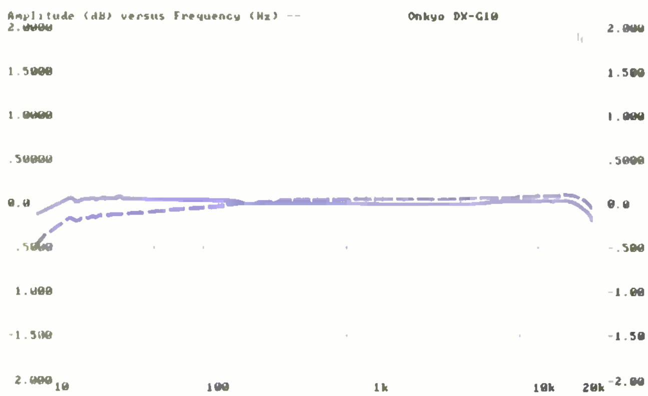


Fig. 1—Frequency response. In this and all subsequent figures, unless otherwise noted, left channel is solid curve and right channel is dashed.

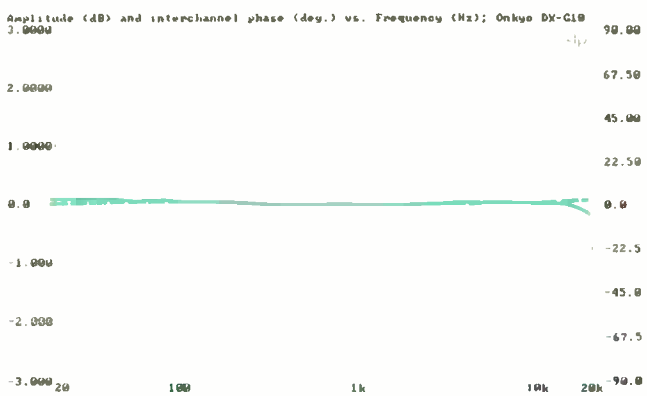


Fig. 2—Comparison of interchannel phase difference (dashed curve) and amplitude response (solid curve). Phase, in degrees, can be read from right-hand scale.

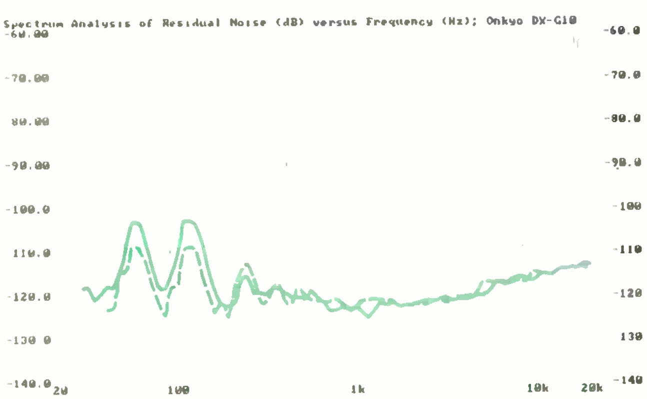


Fig. 3—Residual noise vs. frequency for "quiet" track of CD-1 test disc. Although there are noise peaks at the power-line frequency and its harmonics, they are very low in amplitude.

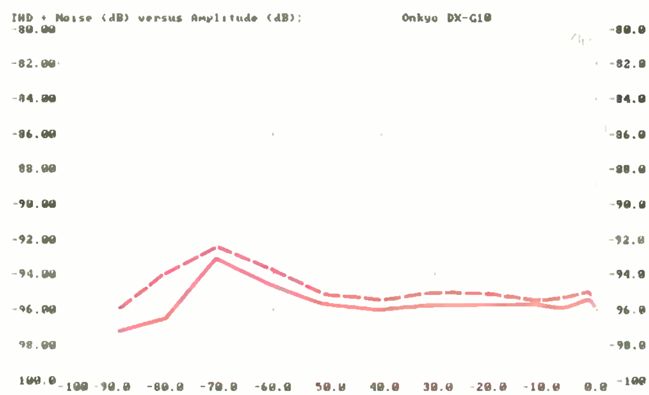


Fig. 4—THD + N vs. signal level.

tions. Pushbuttons beneath the display include "Open/Close" for the disc tray, track advance and reverse, stop, pause, and play. The large "Shuttle Search" knob at the right of the panel resembles a massive volume control, but when turned to the right, high-speed search in a forward direction is initiated; turning the knob to the left obtains the reverse direction. Search speed is determined by how far the user twists this spring-loaded knob in a given direction.

Opening the hinged door along the lower edge of the panel reveals the headphone jack (under the shuttle knob) and a number of control buttons. An "Invert Phase" button is provided for those who are sensitive to polarity reversals in the recording process. There are buttons for specifying a time within a track at which play will begin, for the various modes of repeat play, for index search, and for time-display mode (elapsed or remaining time within a track or total remaining time on the disc). There is even a "Dimmer" button that controls the display area's intensity. Also behind this door are the programming buttons, including the 10 number keys, "Memory," and "Clear." Larger buttons to the right control the volume level at the headphone and variable output jacks. Since the DX-G10 also has fixed-level outputs, its headphone level can be set without affecting output to the main system or a tape deck.

In addition to the variable and fixed analog output terminals, the rear panel has coaxial and optical digital output terminals with a companion slide switch that activates these outputs when needed. The optical output terminal accepts one end of the supplied optical cable, and is of the type that seems to have become the de facto standard.

Access to any point on a disc is superb; you can specify the time where play begins, find index points, or use the shuttle knob.

Measurements

The frequency response of both analog outputs is plotted in Fig. 1. While roll-off at 20 kHz was no more than 0.2 dB for the left channel and 0.1 dB for the right, the right channel exhibited a slight, 0.2-dB roll-off at 20 Hz as well. I can't imagine why this should have been so and suspect that this unusual effect may be the result of an out-of-tolerance component in the analog output stage of my sample. In any event, an imbalance of 0.2 dB between stereo channels at 20 Hz is not something to be very seriously concerned about. In Fig. 2, the plot of the left-channel frequency response is presented along with a plot of interchannel phase difference, which was negligible at all audio frequencies.

A-weighted S/N ratio for this player measured –106.7 dB for the left channel and an almost identical –106.66 dB for the right. Figure 3 is a spectrum analysis of residual noise when playing the "no-signal" track of my CD-1 test disc through the Onkyo unit. As I have explained in previous reports, this graph, as well as the figures just cited for S/N ratio, have nothing to do with the digital performance characteristics of the player; rather, they tell us something about the noise performance of its analog section. Interestingly, despite the extent to which this unit's power-supply section is shielded from its analog audio stages, you can still detect a minute amount of 60-Hz component and its harmonics in Fig. 3. The 60-Hz fundamental and its second harmonic are, however, more than 103 dB below maximum level.

Figure 4 is a plot of THD + N (expressed in dB below maximum output) versus recorded level. Ideally, this should be a straight line. Both channels exhibited a slight rise in THD + N at around 70 dB below maximum record level.

Figure 5 shows how THD + N varied as a function of frequency, for signals recorded at maximum level. Distortion and noise at 1 kHz were almost identical in both channels—0.0018% for the left channel and 0.001% for the right; the results varied only very slightly across the audio frequency spectrum. Rarely have I seen a CD player that did not exhibit a marked rise in THD at 20 kHz, but the Onkyo DX-G10 is one of those rare units. Figure 6 helps to explain why. It is a spectrum analysis of the player's output for a maximum-amplitude test tone at 20 kHz. There is not the slightest evidence of any beat tones, either within the audio band or above it.

Separation between channels was excellent (Fig. 7). Left-to-right channel separation was 111 dB at 1 kHz and 92 dB at 10 kHz, while right-to-left separation was 104.5 dB at 1 kHz and 88.5 dB at 10 kHz. Figure 8 shows a test of linearity, using an undithered, 1-kHz signal at levels ranging from 0 (maximum recorded level) to –90 dB. I noted that both channels began to depart from linearity very slightly (and in opposite directions) below –40 dB. Although their nonlinearity began to increase more rapidly below –70 dB, it was less than 2.0 dB in either channel, even at a level of –90 dB. When I used a low-level, dithered test signal, as shown in Fig. 9, left-channel linearity was nearly perfect—off by no more than about 1.5 dB at –100 dB below maximum recorded level—while right-channel output exhibited a maximum deviation from linearity of only slightly more than 2.0 dB at –100 dB. The fade-to-noise test was conducted for the left channel only (Fig. 10), and the results fully confirm

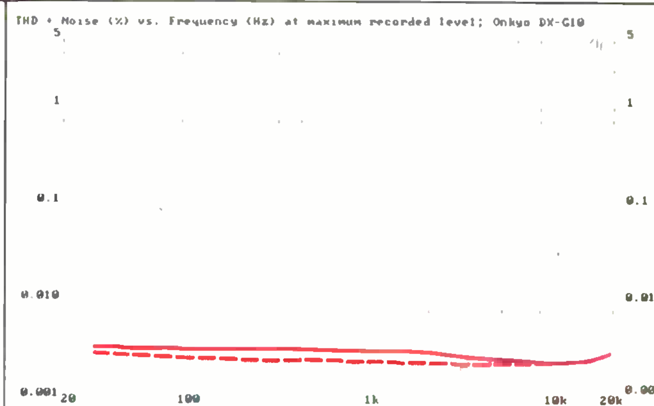


Fig. 5—THD + N vs. frequency, for signal level of 0 dB. The rise at 20 kHz, common in CD players, is minuscule.

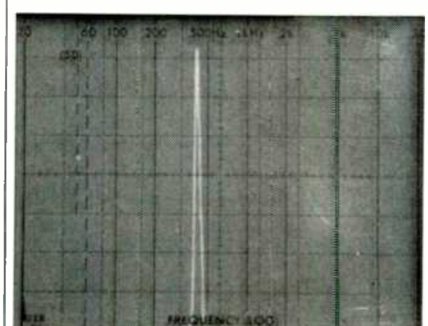


Fig. 6—Spectrum analysis of 20-kHz signal; note the absence of "beat" components. Sweep is linear, from 0 Hz to 50 kHz.

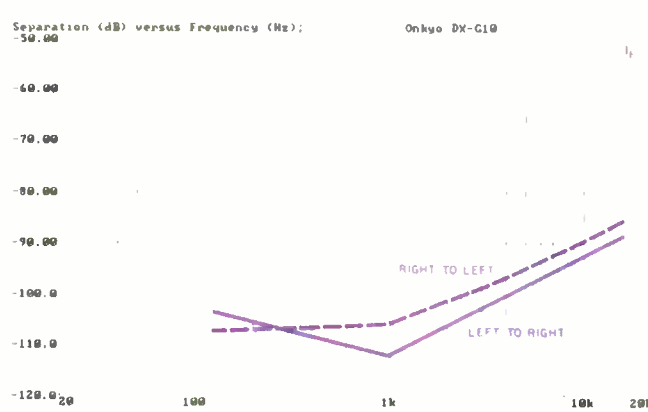


Fig. 7—Interchannel separation.

Rare is the CD player that does not show a marked rise in THD at 20 kHz, but this Onkyo unit is one of those rare exceptions.

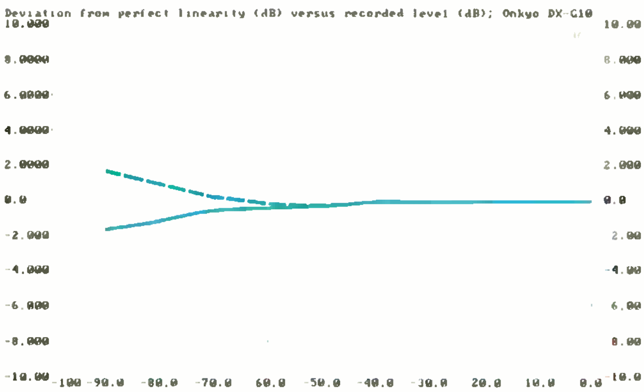


Fig. 8—Deviation from perfect linearity for an undithered, 1-kHz signal at levels from 0 to −90 dB. Linearity was nearly perfect, with maximum deviation not exceeding 2.0 dB in either channel, even at −90 dB.

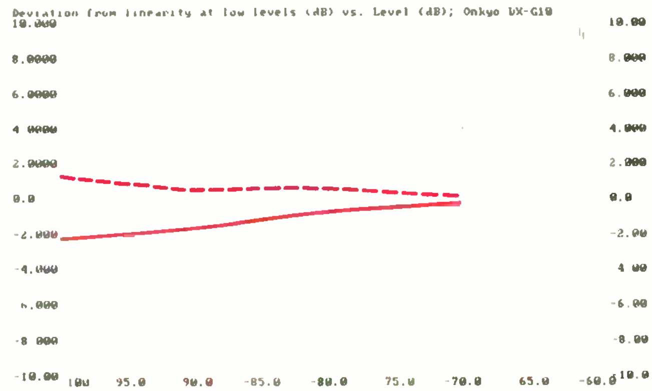


Fig. 9—Linearity deviation for a dithered, 1-kHz signal at levels from −70 to −100 dB.

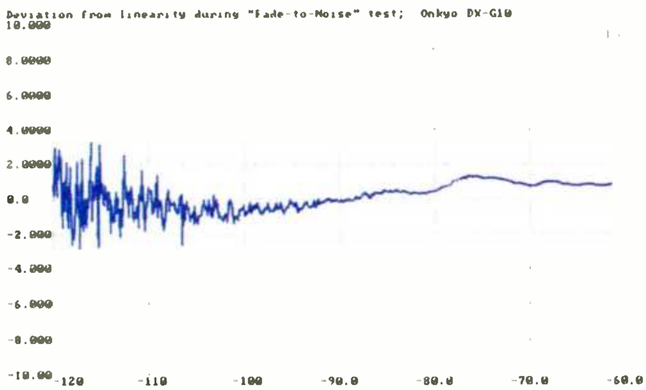


Fig. 10—Linearity deviation for "fade-to-noise" test of dynamic range, using a dithered signal. The DX-G10 tracked the level changes in this test extremely well.

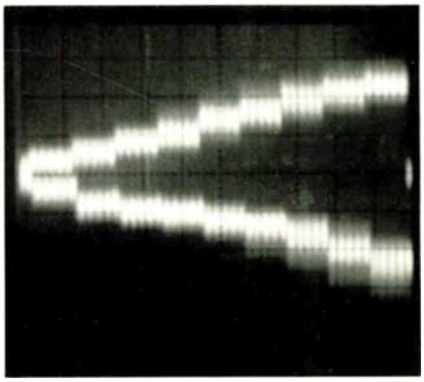


Fig. 11—The monotonicity test also shows this player's excellent low-level linearity.

the earlier conclusion concerning the excellent linearity exhibited by this player. Discrete signals could easily be distinguished from noise, even at levels as low as 110 dB below maximum output.

EIAJ dynamic range measured 93.8 dB for the left channel and 89.7 dB for the right. The monotonicity test signals on the CD-1 disc further verified the DX-G10's excellent low-level linearity (Fig. 11).

I made a few other spot performance measurements before I began my listening tests. SMPTE-IM distortion, at maximum signal level, was 0.00528% on the left channel and 0.00975% for the opposite channel. Frequency accuracy, a measure of the player's master-clock accuracy, was within 0.0159% of perfect. Figure 12 shows how the Onkyo reproduced a 1-kHz square wave, and Fig. 13 depicts a unit pulse as reproduced by the player. It confirms that this unit introduces no signal polarity inversion, unless one chooses to invert polarity by activating the button provided for this.

Use and Listening Tests

For my listening tests, I hooked up the DX-G10 directly to a Hafler XL600 power amplifier (reviewed last month) that had been adjusted for minimum distortion while it was connected to my reference Infinity RS 9 Kappa speakers. In this arrangement, since no preamp was used, I had to connect the CD player's variable outputs to the amplifier. At the time I tested the Onkyo unit, I did not have on hand an amp or preamp with an optical or coaxial digital input, so I was not able to use this player's digital output terminals. Had I been able to do so, however, I would really have been judging only a portion of the Onkyo along with the D/A converter circuit of some other piece of equipment.

The Onkyo DX-G10's low-level linearity proved superb in test after test, including the "fade-to-noise" and monotonicity checks.

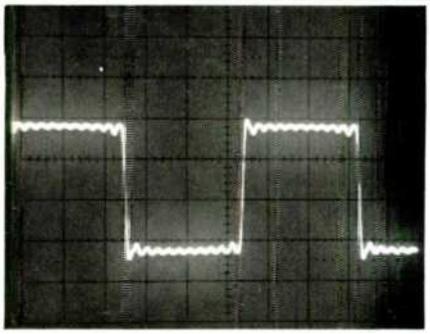


Fig. 12— Reproduction of 1-kHz square wave.

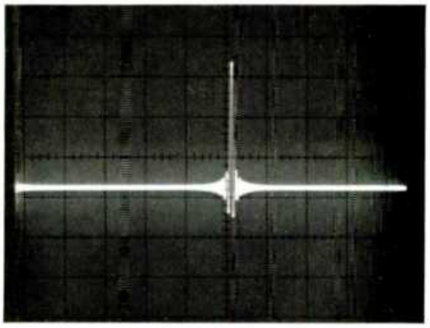


Fig. 13— Single-pulse test.

Among the CDs I listened to were a pair that I had used in evaluating the Hafler amp, namely a disc containing very dynamic classical overtures (Pro Arte's *Light Cavalry*, CDD 402) and a clarinet and piano recording (Delos' *Brahms/ Schumann Soiree*, D/CD 3025). Both discs were digitally mastered and recorded and bear the now-familiar SPARS DDD designation. I was very pleased with the sound delivered by this player/amplifier/speaker combination as a whole.

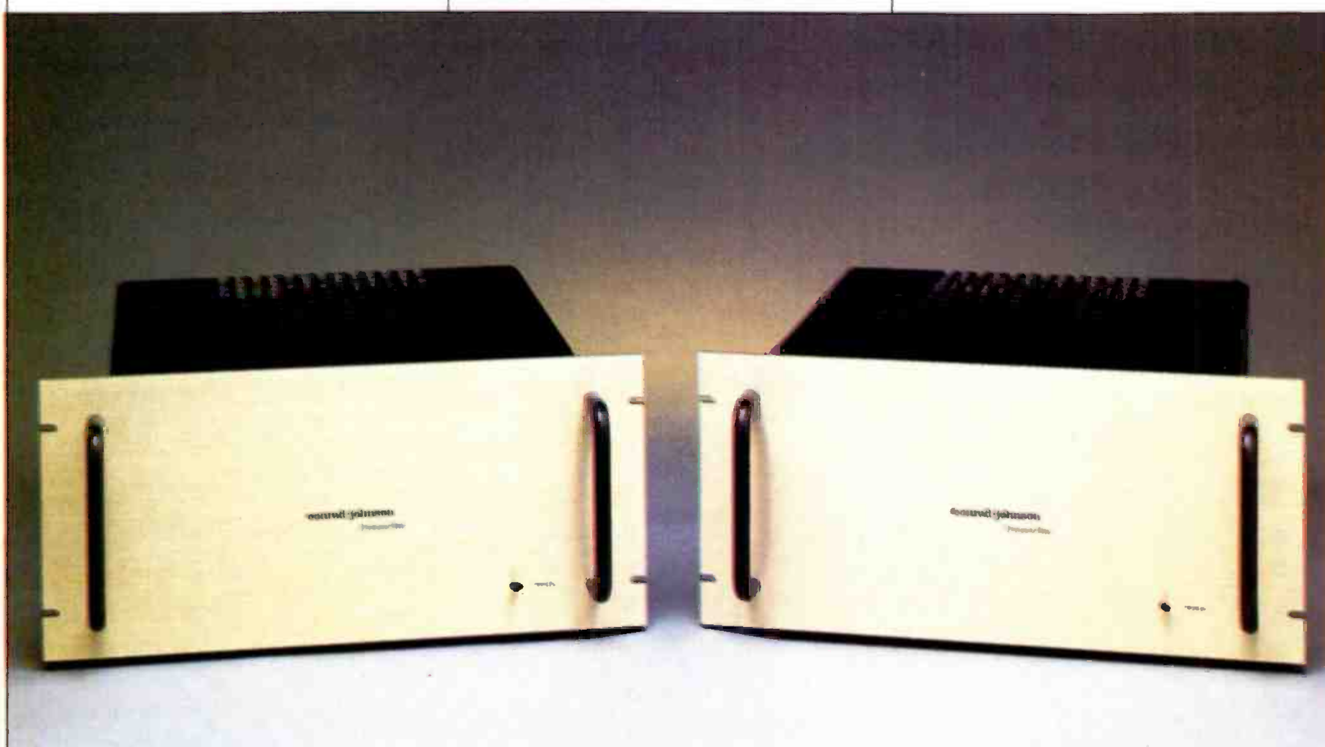
Mechanically, the DX-G10 was a delight to use. Its "Shuttle Search" knob should appeal to those who like to "skip around" in their music, listening to short passages which they are particularly fond of. Access to a given track was so rapid, it was difficult for me to time with a stopwatch. Programming was easy and straightforward. On discs that included index points, I liked being able to move from one to another, even while a track was playing. I also appreciated being able to specify start of play from a given time into a track. In short, I liked everything about this CD player, with the possible exception of its price. At a weight of about 60 pounds, that works out to nearly $42 per pound—or about $93 per kilogram for audio enthusiasts who are into the metric system! Be that as it may, if Onkyo's intent was to show the very best they could do in a high-end CD player, with the DX-G10 they have come very close to succeeding.

Leonard Feldman

8

CONRAD-JOHNSON PREMIER FIVE MONO AMP

Manufacturer's Specifications
Power: 200 watts continuous into 4, 8, or 16 ohms, 30 Hz to 15 kHz, at 1% THD or IM.
Sensitivity: 1 V for full rated power.
Small-Signal Distortion: 0.05% at mid-band.
Frequency Response: 20 Hz to 20 kHz, +0, −0.5 dB.
Hum and Noise: 96 dB below full rated output.
Input Impedance: 100 kilohms.
Dimensions: 19 in. W × 9 in. H × 20½ in. D (48.3 cm × 22.9 cm × 52 cm).
Weight: 81 lbs. (36.8 kg).
Price: $3,000 each.
Company Address: 2800R Dorr Ave., Fairfax, Va. 22031.
(Originally published August 1986)



The conrad-johnson Premier Five is a 200-watt, mono, vacuum-tube power amplifier. It is quite large and heavy, and surely will whet the appetite of any tube-electronics lover. I was very pleasantly surprised, a number of months ago, to have a pair of these beauties arrive on my doorstep. I decided that it would be a good idea to review them, as I had spent a good deal of time listening to two other pairs on Infinity RS IB loudspeakers.

Physically, the Premier Five is built more or less like older tube amplifiers, with a main chassis; a large, thick, rack-width front panel; side pieces, and a top cover. However, instead of using point-to-point wiring between tube sockets and other components, it utilizes a large p.c. board which has most of the interconnections via p.c. traces. The tube sockets are mounted on the p.c. board, and the tubes stick up through holes in the top surface of the chassis. In addition to the holes for the eight output tubes and three front-end tubes, there are holes for a bias pot and a bias-indicator LED for each output tube. The only problem with this construction is that the p.c. board must be partially unwired and swung out if one wishes to replace components on it.

Mounted on the chassis are four large electrolytic capacitors for the power supply, a huge power transformer, and a not-so-huge but still substantial output transformer. On the rear surface of the chassis are a large, four-terminal screw barrier strip appropriate for heavy speaker wire, a plate-current fuse with an LED fuse-out indicator, an RCA signal-input jack, a power-line fuse, and the power cord. The front panel bears a pair of handles and a nonilluminating power switch.

Construction and parts quality on this amp are very good. Reliability is also good, judging from my own experience with the pair under review and with the two pairs owned by Infinity which I had previously auditioned.

Circuit Description

The circuit topology of the Premier Five is similar to that of many older tube designs. The first stage is a grounded-cathode amplifier with two resistors in series from cathode to ground. The signal input is direct-coupled to the grid of the first stage through a 1-kilohm series resistor. Input impedance is set by a 100-kilohm resistor between input and ground. The plate of the first stage is direct-coupled to the grid of the second-stage tube, which is operated as a grounded-plate or cathode follower. The first and second stages use the two halves of a 5751 twin triode tube. Plate-supply voltage to these stages is about 400 V d.c.

The output of the cathode-follower second stage is direct-coupled to the grid of the phase-inverter stage, which is a "long-tailed pair" or differential amplifier. Each tube in the phase-inverter stage is a 6CG7 tube, whose two halves are connected in parallel. The plate outputs of the phase-inverter stage are two equal-amplitude, opposite-phase signals. These are each coupled through four separate capacitors into the grid circuits of four EL34 tubes, which are connected in parallel. Plate-supply voltage for the phase-inverter stage is about 430 V d.c.

This output stage is operated in an ultra-linear connection, with the screen grids of the output tubes fed from taps on the output transformer's primary winding. The B+ for the output stage is about 500 V d.c. Output-stage quiescent current is about 360 to 400 mA.

Output-tube bias is set by a neat arrangement that, to the best of my knowledge, conrad-johnson has used on all their tube power amps. Output tube current is sampled across a 20-ohm resistor between each cathode and ground. An op-amp comparator circuit for each output tube compares the cathode voltage to a fixed reference voltage. Each op-amp comparator output is connected, via an indicating LED, to ground. If a particular cathode voltage is higher than the reference, the output of that comparator goes high, turning on the indicator LED. After a suitable warmup of 15 to 30 minutes, biasing procedure requires one to turn the bias pot for each tube until its LED comes on, and then back it off until the LED just goes out. This is simple and neat, though personally, I would rather have a front-panel plate-current meter and switch to select each tube, along with bias pots accessible on the front panel, as on the Audio Research D150 and the older Marantz Model 9.

Overall negative feedback is taken from the output transformer secondary at the 16-ohm tap, through a 5.1-kilohm series resistor, back to the junction point of the first-stage cathode resistors.

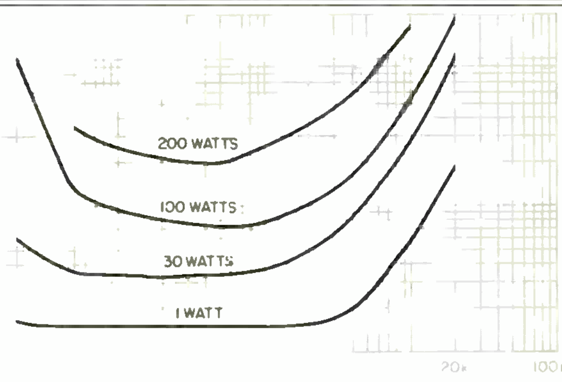


Fig. 1—THD + N vs. frequency for four power levels, with 8-ohm load on 8-ohm tap.

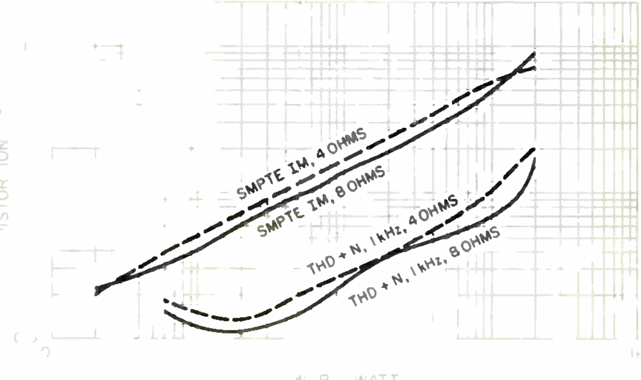


Fig. 2—SMPTE IM (upper curves) and THD + N (lower curves) vs. power for 8-ohm loads on 8-ohm taps and 4-ohm loads on 4-ohm taps. THD + N is for a 1-kHz test signal, with distortion products measured from 400 Hz to 80 kHz.

In the power supply, the high-voltage secondary is full-wave rectified. A capacitor input filter is formed by two 1,300-μF, 350-V capacitors placed in series. Across each of these caps is a 100-kilohm, 2-watt resistor. The resistors equalize the d.c. voltage drops across each of the capacitors, and form a bleeder to discharge the capacitors when the power is turned off—definitely dangerous energy storage here. A series inductor, 0.32 henry at 600 mA, couples

In a super system, these amplifiers are stunningly believable. Even in my less lofty setup, I find them ultimately satisfying.

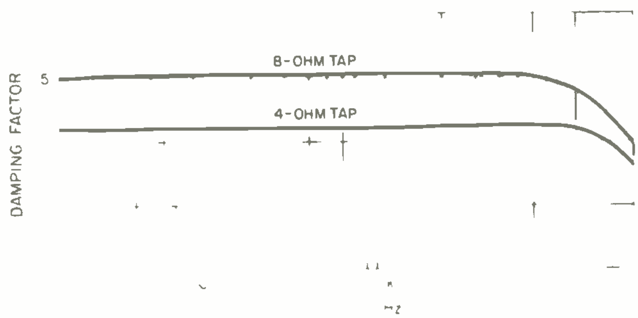


Fig. 3—Damping factor vs. frequency, measured at 8- and 4-ohm taps.

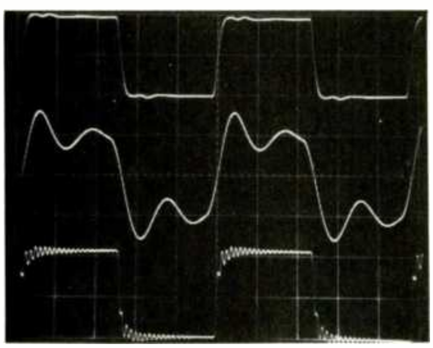


Fig. 4—Response to 10-kHz square wave. Top trace is with 8-ohm resistive load on the 8-ohm tap; middle trace is with 2-μF capacitance across 8-ohm load, and bottom trace is for open circuit (note marginal stability). Scales: Vertical, 5 V/div.; horizontal, 20 μS/div.

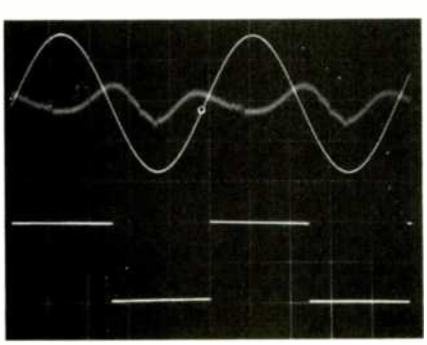


Fig. 5—Top trace: 10-watt, 1-kHz sine wave with distortion products (predominantly even harmonic) shown as residual trace behind it. Bottom trace: 40-Hz square wave, showing excellent low-frequency response. Both signals delivered into 8 ohms. Scales: Vertical, 5 V/div.; horizontal, 5 mS/div.

the peak-rectified d.c. into another capacitor formed by two 3,300-μF, 350-V units in series. Again, 100-kilohm, 2-watt resistors are placed across these capacitors. A parallel combination of two 2-μF, 600-V film capacitors and one 0.15-μF, 630-V film capacitor are placed in parallel with the final electrolytic filter capacitor. The final filtered high voltage is fed to the center tap of the output-transformer primary winding through a 3-ampere fuse that is paralleled by an LED (in series with a limiting resistor) which indicates when the fuse is blown.

The final filtered high voltage also feeds two solid-state zener-follower voltage regulators that supply the regulated voltages of the front-end stages. Across the output of the regulator that feeds the input-amplifier stage are eight 0.15-μF film capacitors. The regulator that feeds the phase-inverter stage is bypassed by a parallel combination of four 1-μF and two 0.15-μF film capacitors.

Another winding on the power transformer is half-wave rectified and filtered, and feeds two separate zener-follower regulators that provide −48 V bias supplies for each half of the output stage. Like the high-voltage supplies, these bias supplies are full of good-quality film bypass capacitors.

A third secondary winding on the power transformer is full-wave bridge rectified and capacitor-filtered to feed smoothed d.c. to the heaters of the front-end tubes. A fourth secondary winding is half-wave rectified to a plus-and-minus supply which provides the supply and reference voltages for the op-amp's bias indicator circuits. A fifth (and final) secondary winding provides 12.6 V a.c. to power the heaters of the output tubes.

To sum up the Premier Five's circuitry: The amplifier circuit itself is fairly straightforward, with the exception of the cathode-follower buffer between the first amplifier stage and the long-tailed-pair phase inverter. The power supply has a lot more filter storage capacitance than older tube-amplifier designs. This, in conjunction with the voltage regulators powering the front-end circuitry, most likely helps keep things more solid—especially under large-signal conditions. The liberal use of low dielectric-absorption, film bypass capacitors throughout the power supply probably helps sonic performance considerably.

Measurements

The first step in measuring the Premier Fives' performance was to rebias the output tubes to the correct idling current at an a.c. line voltage of 120 V. This current, by the way, is 45 to 50 mA. In my house, the line voltage is more like 112 to 114 V with the amps on. Before I rebiased the amps, I measured the mid-band power, at the onset of clipping, with 112 V from the power line. This worked out to about 180 watts.

Voltage gain, with an 8-ohm load on the 8-ohm tap, was 36× or 31.1 dB, which is some 5 dB higher than the usual power-amp gain of 26 dB. For the 4-ohm tap, gain was 26.5× or 28 dB. IHF sensitivity for 1 watt out into 8 ohms was 78.5 mV.

Figure 1 shows THD + N versus power and frequency, for 8-ohm loads on the 8-ohm taps. As can be seen, distortion rises above 1 to 2 kHz for all power levels shown. At higher power levels, distortion also rises at low frequencies.

CD player
all of the music,
nd bytes of it.

This higher resolution makes an audibly dramatic difference in the musicality of CD reproduction. To achieve this result, the analog audio circuits in the GCD-575 were based on the same proprietary high speed linear amplifiers used in Adcom's GFP-555 preamplifier, universally recognized for its outstanding musical integrity.

No other CD player at any price uses these superb audio components.

Digital Sound At Its Best

Adcom's selectable analog frequency/phase contour circuit enhances the musicality of CD's which have been poorly mixed, or digitally over equalized. Subtly contoured by the AFPC, many of these CDs become more listenable, with much of the fatiguing harshness and "glare" reduced. In addition, the stereo image and sound stage becomes more focused allowing for a more natural sonic presentation.

(Over please)

The Adcom GCD-575

Details You Can Hear

Importantly, Adcom's CD player is designed with a low output impedance (100 Ohms) so that it can operate up to its maximum capability with a wide variety of associated equipment. It is not only compatible with virtually all input stages of amplifiers, preamplifiers, tuner/preamplifiers, etc., but also permits the use of longer interconnecting audio cables, when required, with minimal signal deterioration.

Additionally the GCD-575 is supplied with a high quality, low-loss audio cable to prevent the sonic smear that conventional audio cables tend to cause. The use of this special cable and the 100 Ohm output impedance permits the GCD-575 to be used with Adcom's SLC-505 passive straight line controller. If no other source equipment will be used, the variable output (front panel controlled) can be used directly into your power amplifier, bypassing the preamplifier circuits normally required by other CD players.

A multi-winding power transformer, connected to three separate tightly regulated power supplies for the audio, digital and display circuitry, insures isolation of the different functions and optimal operation of each without interference.

The four special heavy feet installed on the GCD-575 are reversible metal castings. On one side, the flat surface insures a wide contact area. The reverse side is cast with built-in "Iso-points" which, when used in a three-foot configuration, operates as a "tripod" support system.

A special polarity-inverting switch permits you to reverse the normal positive polarity to negative (inverted) polarity. This corrects playback of CDs in which the polarity was incorrectly recorded (inverted), or for use in systems in which one of the components causes a reversal of correct polarity.

Full Function Remote Control

Specifications

Frequency Response: 5Hz - 20kHz, +0.1, −0.5dB

Signal-to-noise Ratio: 105dB

Dynamic Range: 98dB

THD: 0.0025%

IMD (70Hz difference): @ 5kHz 0.00018%

Channel Separation (1kHz): 95dB

Interchannel Phase Shift: @ 20kHz Less than 1.8°

Output Impedance: Fixed 100Ω/ Variable 100Ω/Digital 75Ω

Output Level: Fixed 2.5V RMS Variable Greater Than 4.5V RMS Digital 0.5V peak-to-peak

Sampling Rate: 176.4kHz

Quantized Bits: 16-bit linear

Power: 120VAC/60Hz (Available in 220/240V, 50Hz)

Dimensions: 17″ (430mm)W × 11-1/4″ (285mm)D × 3-7/16″ (87mm)H

Weight: 12 lbs. (5.5 kg.)

Optional: Model RM-3 rack mount adaptors. Available with white front panel.

Specifications subject to change without notice.

More Features For Better Value

Other features include a full function remote-control system with random access track capability; low group-delay digital and analog filters; triple-beam laser format; a direct digital output; playback of 3-inch discs without an adaptor; and a very-high-quality headphone output.

The GCD-575's advanced facilities include:

- Programming of up to 24 tracks
- Programming of any phrase
- Audible fast forward and reverse
- Adjustable introscan
- Auto space

Display functions include:

- Elapsed time on track or disc
- Time remaining on track or disc
- Programmed tracks
- Track being played
- Number of tracks up to 20

Why Should You Listen To Us?

Over the years, Adcom has earned a reputation for delivering superb performance at a modest price. The GCD-575 keeps faith with this tradition.

Once again, Adcom clears an innovative path through the jungle of confusing claims about "digital" sound, and provides a logical and direct path to musical purity.

If you've been waiting for a CD player which faithfully reproduces all of the music, not just bits and bytes of it, you'll want to visit your nearest authorized Adcom dealer right now...because while it may be true that all good things come to those who wait, you've waited long enough for a CD player this good.

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Pointe Claire, Quebec H9R 4X5

Enter No. 2 on Reader Service Card

I wish I could quantify why the Premier Fives sound so good, but I can't. So I turn off my rational side and just enjoy the music.

At 20 Hz, the amp could not produce 200 watts output due to output-transformer saturation.

Figure 2 shows THD + N (measured from 400 Hz to 80 kHz, for a 1-kHz test signal) and SMPTE-IM distortion, for 4- and 8-ohm loads on their respective taps. The amp's behavior on the 16-ohm tap is about like that on the 4-ohm tap. For some reason that I can't figure out, the 8-ohm THD at 200 watts shows onset of clipping, but distortion with 8-ohm loading is lower below 200 watts than with either 4- or 16-ohm loading. With either 4- or 16-ohm loading, the THD residue produced at 200 watts output does not exhibit onset of clipping.

Damping factor versus frequency is shown in Fig. 3 for the 4- and 8-ohm taps. Damping factor is higher yet on the 16-ohm tap, because the feedback is taken from this tap on the secondary of the output transformer.

Rise- and fall-times into an 8-ohm load were 3.5 μS at ± 5 V output. Oscilloscope photos for various conditions are shown in Figs. 4 and 5. The top trace of Fig. 4 is for a 10-kHz square wave into 8 ohms, driven from the 8-ohm tap. The middle trace is with 2 μF of capacitance paralleled with the 8-ohm resistive load. The bottom trace is for an open circuit; stability here is marginal. Square-wave performance on the other taps is similar, but not the same in terms of ringing and overshoot. The top trace of Fig. 5 shows the nature of the harmonic residue, which is predominantly even harmonic, at 10 watts output at 1 kHz. The lower trace shows excellent low-frequency response with a 40-Hz square wave.

Frequency response at 1 watt output, for 8-ohm loading, is shown in Fig. 6. The IHF signal-to-noise ratio was found to be -82 dB. IHF dynamic headroom measured 218 watts or 0.37 dB, and IHF clipping headroom was 205 watts or 0.11 dB; both were measured with 8-ohm loading and 120-V a.c. line input.

Peak current into a 0.1-ohm load on the 4-ohm tap, using the IHF dynamic-headroom test signal of 20 mS on and 480 mS off at 1 kHz, yielded ± 22 amperes before visible distortion occurred.

Summing up on measurements: The Premier Five tube amplifier has higher distortion figures near full power than most solid-state power amplifiers, although at low to medium power levels it is satisfactorily low. High-frequency stability might be a problem with a load that presents a high impedance at ultrasonic frequencies. My only actual experience as evidence of this occurred when driving an Infinity IRS speaker, with its tweeter disconnected temporarily for test purposes. A buzz in the midrange drivers suggested that the amp was oscillating under this abnormal condition.

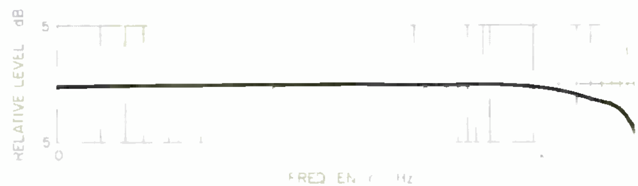

Fig. 6—Frequency response at 1 watt output into 8-ohm load.

Use and Listening Tests

A comment on my personal preference or bias is in order here: Some of my reviews may give the impression that I don't care for solid-state gear and that I prefer tube equipment. I would like to clarify this. Good tube equipment, for me, simply re-creates (or creates, if you will) a more believable, emotionally involving musical experience. Where tube equipment gets it more right is in the areas of spaciousness, depth of image in the sound field, and instrument tonality. Thus far, solid-state gear doesn't quite measure up in my opinion, although the gap is narrowing. I do like to use solid-state equipment because of its long-term reliability, stability of characteristics, lower power consumption, etc. With this said, on to my evaluation of the Premier Five amplifiers' sonics.

As previously mentioned, I had the opportunity to hear two other pairs of Premier Fives, in the Infinity Systems sound room, on RS IB and IRS loudspeakers. The Infinity system uses a Mitchell A. Cotter turntable with a Goldmund tonearm and Koetsu Onyx cartridge. The resident preamplifier is an Audio Research SP10. With Premier Fives driving the midrange and tweeter sections of a pair of RS IBs, the sound of this system is very good indeed. I have listened to a good number of transistor amplifiers on this system; in comparison to the Premier Fives, they all sound variously less dimensional, more irritating, and ultimately less musical to my ears.

I have heard the personal system of Arnold Nudell, Infinity's president, a number of times. The signal source in this setup consists of an Otari professional open-reel recorder, playing low-generation copies of master tapes, with or without transformerless Dolby A NR units. The signal from the Otari is fed, via a dual 500-ohm volume control, into the bass amplifier and crossovers of an IRS speaker system. Nudell uses Premier Fives to drive the midrange and tweeter panels. Reproduction is stunningly believable, which tells me that the Premier Fives are incredible amplifiers.

In my less lofty home listening environment (using an Infinity air-bearing turntable, Koetsu's new EMC-1B cartridge, Infinity RS IIB speakers, and Stax SR-X/Mk3 headphones), I have found the Premier Fives to be ultimately satisfying. I keep trying other amplifiers and when I go back to the Fives, my reaction is, "Ahhh, all right!" Even my super-critical associate, Geoff Cook, concedes that they are "pretty good amps." The only other power amplifiers that have satisfied me as these do are the Marantz 9s, which sound a little softer and sweeter in the high end and not quite as solid in the bass. Of course, the 9s are no longer commercially available, whereas the Fives are. I like the Premier Fives very much, and would recommend that anyone who can afford them give them a serious audition.

As a concluding point, I wish I could quantify with some measurements why the Premier Fives sound so good. As a measurer, I don't yet have a clue. This is frustrating, and I hope to ultimately find out why. In the meantime, I have no trouble turning off the rationalist, the language-oriented, measurer part of me, turning on my ears, the ultimate measurer, and enjoying the music. *Bascom H. King*

9

THRESHOLD SA/1 MONO AMP

Manufacturer's Specifications
Power Output: 160 watts continuous into 8-ohm loads, 20 Hz to 20 kHz.
THD: Less than 0.1%.
Frequency Response: 7 Hz to 100 kHz, for −3 dB points.
Slew Rate: 50 V/μS.
Output Current Capability: 40 amperes continuous, 60 amperes peak.
Input Impedance: Less than 0.03 ohm, 20 Hz to 20 kHz.
Gain: 26.6 dB.
S/N Ratio: 100 dB, unweighted, re: rated output.
Load Rating: Operation permitted into any load, limited only by power-supply fuses and thermal protection.
Dimensions: 19 in. W × 9.65 in. H (including feet) × 17.25 in. D (48.3 cm × 24.5 cm × 43.8 cm); face-plate, 8.75 in. H.
Weight: 78.5 lbs. (35.6 kg).
Price: $3,950 each.
Company Address: 12919 Earhart Ave., Auburn, Cal. 95603.
(Originally published January 1987)

Editor's Note: The SA/1e, the latest version of the SA/1, uses optical coupling of the bias current. According to Threshold, this results in increased output current and a 25% reduction in measurable distortion.—*E.P.*

Photograph: Jay Brenner

To justify a price of $3,950 for a monophonic power amplifier (or even a stereo one), the company brave enough to offer such a product must offer a truly unique design. Nor is innovative design enough: To convince even the most affluent of audiophiles to spend $7,900 for a pair, such an amplifier had better be audibly very superior to the competition. In their SA/1 amplifier, Threshold has, in fact, brought their "Stasis" concept to its ultimate realization. The overall design is brilliant, and the sound quality is so transparent and clean that few superlatives exist to do it justice. All of this does not explain why Threshold needs to ask so staggering a price for the SA/1. The explanation has to do with the difference between a true Class-A design (which the SA/1 is) and the myriad pseudo-Class-A designs which have appeared during the last few years. The latter purport to provide all the benefits of Class A without the most obvious disadvantage: The high currents drawn even when no signal is applied. Most true Class-A, solid-state designs which I have evaluated in the past could boast no more than a 20 to 30-watt power rating because of the continuous current requirement. The SA/1 is conservatively rated at 160 watts continuous power output. To reach this power level, Threshold uses no fewer than *40* output transistors (20 NPN-PNP pairs) in what looks, at first, like a conventional complementary-symmetry output-stage configuration. Upon closer examination, though, you will discover that the output stage is anything but conventional.

Stasis Technology

Figure 1 is a much-simplified diagram of the Stasis principle. Threshold provides a very clear explanation of just how it works in the SA/1 amplifier (and others in the Stasis series), and how this amp is different from any other. Since the manufacturer does a better job of explaining the circuit than I could myself, some of that explanation follows:

"The output stage of a Stasis amplifier is so accurate in its operation that no corrective (feedback) loop needs to be employed around the amplifier system to impose 'compensating' inverse inaccuracies.

"Threshold Stasis amplifiers do not attempt to fool a nonlinear system into exhibiting linear results through imposition of a corrective distortion loop around the amplifier system. Instead, they are able to maintain the transistors themselves under such unvarying conditions of voltage and current that their gain characteristics become virtually perfect so that no corrections are required.

"The Stasis operating principle is based upon a simple and incontrovertible fact: The current through a load is determined by the voltage across it. If the voltage across the load is free of distortion, the current will be distortion-free too and so will the power delivered. Threshold Stasis amplifiers embody a patented binary output stage in which an extremely linear voltage amplifier, operated in pure Class A, determines signal accuracy by supplying the voltage across the load. The voltage amplifier is connected *directly* to the load and to a powerful current mirror which is also connected to the load and to the voltage amplifier. It is the current mirror's 'bootstrap' that supplies the current—or power—through the load.

"As the voltage amplifier has to do no amplification 'work,' its gain transistors can be held in a stasis—or virtually unchanging—condition relative to voltage and current. Without being subjected to fluctuating voltage and current, the devices of the Stasis section maintain a uniform gain factor and, by definition, linearity and freedom from distortion. Because the output impedance of the Stasis section voltage amplifier is very low relative to the output impedance of the current bootstrap, it is able to dominate the performance of the tandem system. In this way, the accuracy of the amplifier is that of the extremely linear and precise Stasis section."

Additional Circuitry

The input stage of the SA/1 consists of four matched N-channel J-FETs. One pair is arranged as a matched differential input pair which is operated at a constant voltage through cascode shielding by the second pair. Secondary cascoding is also employed, so that the regulation associated with the cascode operation is applied to the input J-FETs twice. This results in better isolation of the audio-signal circuits from power-supply dynamics and signal-induced voltage variations. The extremely high active impedance of the input J-FETs, along with their high linearity and low transconductance, offers a variety of benefits. These include reduced interaction with the signal source, greater power-supply rejection ratio, and the need for only a very minimal amount of local feedback.

The output stage of the SA/1 employs fast power transistors, each rated at 200 V, 20 amperes. This results in a greatly extended safe area of operation; output transistors are used at a small fraction of their power capabilities.

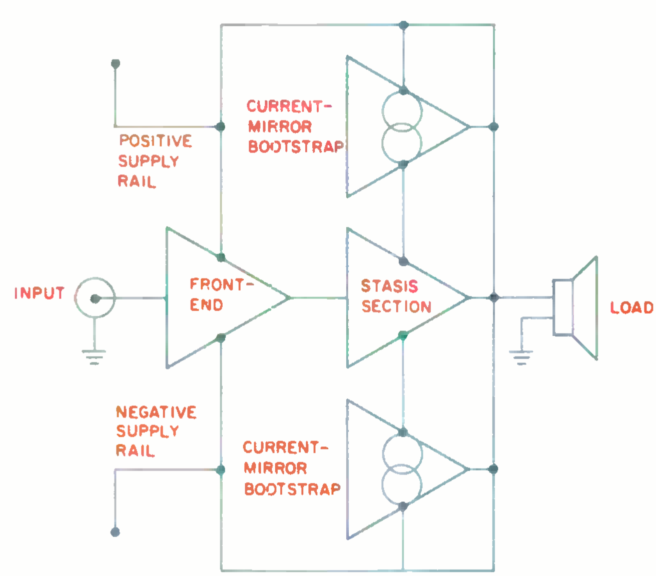

Fig. 1—Simplified block diagram of Stasis circuitry.

To reach its rated power level of 160 watts, the SA/1 uses no fewer than *forty* output transistors in a configuration that is anything but conventional.

Another benefit of this conservative design is that it is not necessary to use active and often audible protection mechanisms to safeguard the output stage or to introduce any fusing between the output transistors and the loudspeaker load.

Eliminating this fuse lowers the source impedance of the output stage, thereby increasing the amplifier's damping factor. In conventional designs, high damping factors are directly related to the use of large amounts of overall loop feedback. What is often overlooked in such designs is the fact that at high frequencies, feedback is often reduced to very low amounts because of the need to maintain amplifier stability. The result: Lower damping factors at high frequencies. Threshold maintains that high damping factors are needed even at high frequencies, where, for example, an electrostatic loudspeaker's impedance may drop to 1 or even 0.5 ohm. A damping factor of only 10 at those frequencies would mean the amplifier's impedance would be 0.8 ohm, low enough to cause a power loss of more than 3 dB into the electrostatic's ultra-low load impedance. A high damping factor at higher frequencies will even benefit cone loudspeakers, as it results in lower distortion and improved transient response. Since the SA/1 does not derive its high damping factor from feedback and does not use an isolation inductor at the output, a high damping-factor value is maintained across the entire audio frequency band.

The SA/1 employs a huge, toroidally-wound power transformer. Its output is fed to 35-ampere rectifying diodes and is smoothed to d.c. by electrolytic capacitors totaling 120,000 µF of capacitance. (One could almost start expressing that much capacitance in farads, i.e., 0.12 F!) The supply voltage undergoes additional filtering before reaching the actual audio circuits. A combination of electrolytic and high-value film capacitors is used for this purpose.

Physical Description

Gold-plated, machined, Teflon-insulated input connectors are used in the SA/1. High-current, five-way binding posts have contact surfaces that are also gold-plated. Circuit boards are made of military-grade glass-epoxy with paths plated in gold over nickel. All internal connections are hand soldered. I found metal-film and wire-wound resistors throughout, along with film and silver-mica capacitors in the actual signal-path circuitry.

A large, well-calibrated peak-reading meter dominates the elegant front panel of the SA/1. The only control is the on/off rocker switch, which is, in reality, a high-capacity circuit breaker. Input and output terminals, as well as line and "+" and "−" B-supply fuses, are accessed from the

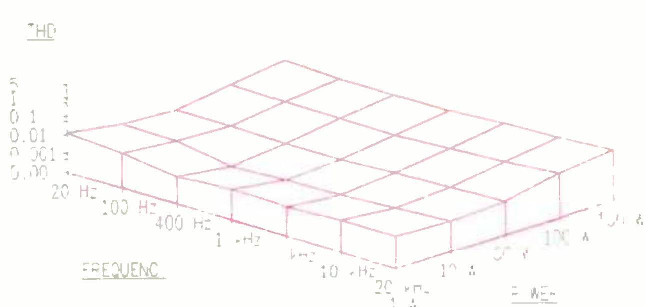

Fig. 2—THD vs. power and frequency, 8-ohm load. THD scale is logarithmic.

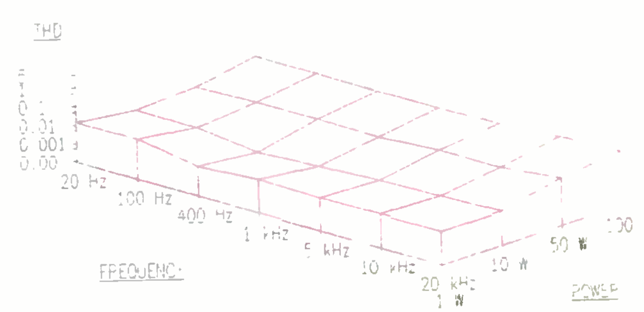

Fig. 3—Same as Fig. 2, with 4-ohm load.

Just a few measurements confirm the fact that a good-sounding amp like this one will measure well too. We all know the reverse isn't always true!

rear panel. Dual banana plugs are supplied for wiring up speaker cables so that they can be easily plugged into the amplifier's five-way output terminals. Spare fuses are also included with each SA/1.

Measurements

It took only a few measurements on my part to confirm the fact that this amp, which had sounded so good in the earlier listening tests, measured well, too. (The reverse, as we all know, is not always true!) At mid-frequencies, the SA/1 delivered 190 watts of continuous power into 8-ohm loads for its rated THD value of 0.1%. It did almost as well at the audio frequency extremes, where just over 180 watts resulted in a THD reading of 0.1%.

I've always had problems measuring THD for power amps at very low output levels. When you get down to 1 watt or so, residual noise and hum start entering into the distortion measurement. Since I don't like to doctor these measurements with band-pass filters, which often obscure other problems, I end up having to use a spectrum analyzer to isolate the actual THD from the hum and noise. Even at that, unless the analyzer has a very wide dynamic range, it's not always possible to isolate an accurate THD reading from the rest of the unwanted components that show up on the analyzer's screen. All of this is by way of telling you that the Threshold SA/1 yielded a THD reading of only 0.005% at 1 watt output into 8 ohms *without* my having to resort to filters of any kind or to interpret results on a spectrum analyzer. My good old distortion analyzer managed to read this low figure all by itself! In fact, at 10 watts output, the THD sank even lower—down to an incredibly low 0.0015%! When I pushed the amplifier up to its rated output of 160 watts, THD rose to 0.02% at mid-frequencies and 0.025% at 20 Hz. Threshold's 0.1% rating was evidently intended to take care of the slight rise in THD that did occur at the high-frequency end of the spectrum. Still, at 20 kHz, I read only 0.06% THD, and the 0.1% level was not reached until the amp was delivering 182 watts at 20 Hz, 190 watts at 1 kHz, and 180 watts at 20 kHz. SMPTE-IM distortion measured exactly 0.1% at rated output but was only 0.01% at 100 watts output. As for CCIF-IM or IHF-IM distortion, if there was any in this amplifier it was too low for my test instruments or spectrum analyzer to detect. A "three-dimensional" diagram, Fig. 2, illustrates how THD varied with power output and frequency.

I made a complete set of measurements using 4-ohm loads as well, even though Threshold does not provide an official power rating for this impedance. Into such a load, the amplifier delivered a power output of 335 watts at 1 kHz and 324 watts at 20 Hz for a THD of 0.1%. The test conditions for Fig. 3 were similar to those for Fig. 2, except that the load impedance was 4 ohms instead of 8.

Although I did perform some measurements of power output at 2 ohms, it soon became clear that if I tried to push the amplifier to its limits with that type of load, I would blow its fuses. Most likely, I would also exceed the current capacity of the voltage regulator which I normally use to maintain a steady line voltage of 120 V when making such critical measurements. Accordingly, for data involving 2-ohm loads I'm going to rely upon curves supplied by Nelson Pass, the well-known designer and president of Threshold. I'm told not only that Mr. Pass was responsible for this superb amplifier design but that he personally supervises its production and makes certain that each unit leaving the factory measures up to his standards. I think it's safe to accept his curves as being correct, and so they are included here as Fig. 4. His measurements show that even when driving a 2-ohm load, this remarkable amplifier delivers around 600 watts at mid-frequencies, and over 500 watts at 20 kHz, for a THD level of around 0.15%. I don't know of many other power amplifiers at any price that can boast this power output with such a low-impedance load presented to them.

Amplifier gain of my particular sample was a bit higher than the 26.6 dB claimed; I measured it to be 29.5 dB from input to output, referring again to an 8-ohm load. Frequency response was off by 1.0 dB at 14 Hz and 55 kHz, while the −3 dB points occurred at 7 Hz and 100 kHz, exactly as claimed. Damping factor measured exactly 500 at 50 Hz. In fact, it was probably a bit higher, since I did have to use a couple of feet of Monster Cable between the output terminals and the measuring instruments, and this added a tiny

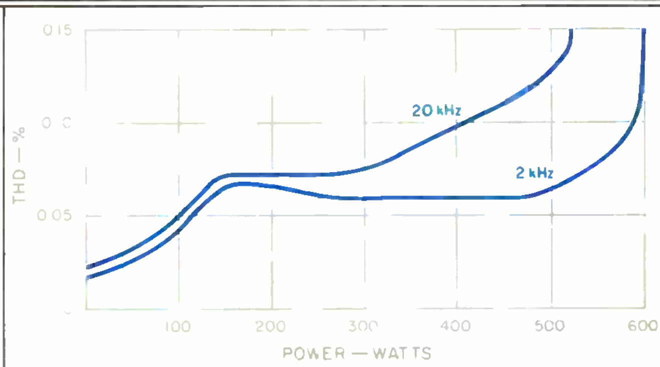

Fig. 4—THD vs. power at 2 and 20 kHz, 2-ohm load. Data was supplied by manufacturer; see text.

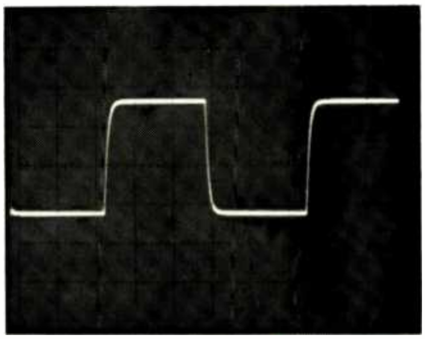

Fig. 5—Response to 20-kHz square wave at 80 watts into 8 ohms.

"State of the art" is a bit shopworn, but there's no other succinct phrase to describe the utterly clean, open sound of this amp.

fraction of an ohm to the "looking back" impedance of the amplifier. Signal-to-noise ratio, referred to 1 watt output, measured 90 dB. (No wonder hum and noise did not affect the THD readings at low power output levels.) Translating this to an S/N value below rated output (the way Threshold quotes S/N), I come up with a figure of 112 dB. Incidentally, all of these measurements are unweighted, since that's the way Threshold quotes their noise figure. If I were to add an A-weighting network (the standard method approved in the EIA Amplifier Measurement Methods Standard), the S/N figure would be even higher.

The SA/1 was not designed to provide much dynamic headroom above its continuous rated power level; I measured 0.6 dB. However, I have never regarded dynamic headroom as a measure of quality. It's nice to have lots of it if you're playing music that has a wide dynamic range through a low-power amplifier into speakers of fairly low efficiency. When you start out with an amplifier that can deliver as much *continuous* power as this one does, extra dynamic headroom becomes rather academic!

To illustrate the superb rise-time and slew rate of this amplifier, I fed a 20-kHz square wave into it. The resulting output, at half rated power into 8 ohms, is shown in Fig. 5, and like everything else about this unit, it is outstanding. I've seen 1-kHz square waves from some amplifiers that don't look this good!

Use and Listening Tests

Most of the bench measurements were made on a single sample of the SA/1; I then set up two samples for extended listening tests, which have been going on for more than a week as I write this. I'll be sorry to part with these amplifiers—and not just because it's going to be difficult to lift them back into their cartons. I know that the phrase "state of the art" has become a bit shopworn, but there's no other succinct way I can describe the utterly clean, open sound of the SA/1. Not having a variety of speaker loads with which to try it out, I queried some of my dealer friends who have had experience with the Stasis series of amps, including this one; they tell me they have yet to encounter an unusual speaker load that would present problems.

Do I have any negative comments to offer? Yes, just one: It's too bad an amplifier that performs this well must cost as much as it does. The high price of this audio masterpiece necessarily limits the number of people who can actually own and enjoy it. And, of course, if you do own a pair, you've got to be prepared to run up your electricity bill somewhat, as each unit does draw around 3 amperes from the line *at all times*. That's one of the penalties you pay for Class-A operation. On the other hand, if you have enough money to buy a pair of Threshold SA/1 amplifiers, the few cents a day extra that you'll be giving to your utility company is not likely to bother you at all.

Leonard Feldman

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YBA$_1$ DUAL MONO AMPLIFIER

Manufacturer's Specifications

Rated Output (From 115-V Line Power): Continuous power, 85 watts per channel into 8 ohms, 170 watts per channel into 4 ohms; pulse power, 1.8 kilowatts per channel into 0.7 ohm.

THD: Less than 0.09% from 20 Hz to 20 kHz, at 20 watts per channel.

Frequency Response: 5 Hz to 80 kHz, +0, −3 dB.

Rise-Time: 3 μS at 10 kHz.

S/N: Greater than 100 dB, unweighted.

Input Impedance: 27 kilohms.

Input Sensitivity: 1.1 V for full rated output.

Damping Factor: Greater than 800 at 100 Hz.

Power Consumption: Quiescent, 100 VA; 1,000 VA at full rated power, both channels driven.

Line Voltage: 115 or 230 V (factory set), 50 to 60 Hz.

Dimensions: $16^{15}/_{16}$ in. W × $5^{3}/_{16}$ in. H × 13 in. D (43 cm × 13.2 cm × 33 cm).

Weight: 46½ lbs. (21 kg).

Price: $6,000.

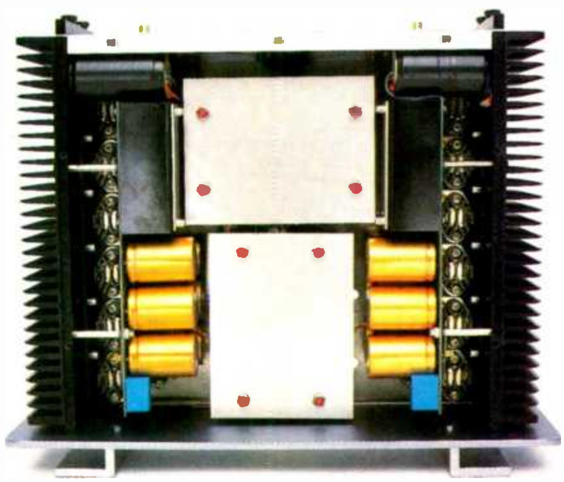
Company Address: c/o Sumiko, P.O. Box 5046, Berkeley, Cal. 94705.

(Originally published January 1989)

The YBA_1 is the largest of three solid-state power amplifiers manufactured by Phlox Electronique of France and imported and distributed in the U.S. by Sumiko; three companion preamps are also available. YBA stands for the initials of Yves-Bernard André, the designer of these six electronic units. Also designed by André, under the Vecteur name, are a turntable and loudspeaker.

Physically, the YBA_1 is about normal in its dimensions for its power rating, but, weighing in at 46½ pounds, the system is definitely heavy for its size. This dual-mono unit is solid, very nice to look at, and built like the proverbial brick outhouse. A number of unusual ideas and great attention to detail are incorporated in the YBA_1's engineering, making it clear that the designer intended this amp to function properly for a long time. Some of the design features include: Topology and positioning of components to limit their interaction, orientation of each of the amp's two-terminal passive components in the best-sounding direction (i.e., in which direction a part should be aimed with regard to signal flow), common-ground star network, minimum internal cable and signal-path length from input to output, and the use of noble nonmagnetic materials with oriented crystal structure, internal wiring of long-crystal oxygen-free copper, power transformers with two C cores, and custom-made passive components. Still more design features will become apparent in subsequent discussions of this amplifier's circuit design and performance.

The amp is built with the two side heat-sinks as its main anchoring pieces. Separate front and rear panels and top and bottom covers bolt to the heat-sinks to form the complete unit. A power switch is hidden behind the front panel's left-side handle, and a single red LED below the logo in the panel's center indicates power on/off. On the rear panel are two pairs of female sockets for speaker cable connection via specially supplied gold mating plugs. Also found on the back of the unit are two special-design phono input connectors, two test-tip jacks for setting quiescent idling current, two circuit breakers, and a combined three-wire power connector/fuse-holder. Two separate C-core power transformers, mounted at right angles to each other and isolated by rubber bushings and grommets for minimum mechanical hum, take up about half of the interior space. Filter capacitors and rectifier diodes are on the unit's p.c. boards, which are mounted parallel to the heat-sinks with standoffs. Driver and output devices are on L-shaped ledges that bolt to the inside surfaces of the heat-sinks. The tops of the TO-3 devices (six per channel) have cast finned radiators attached for added cooling and mechanical damping. Small p.c. boards holding some of the bias-regulator components are mounted between the radiator fins, atop one TO-3 device in each channel. A rather sizable plastic box taking up slightly less than half of each amp's p.c. board, houses each channel's front-end circuitry. These boxes are filled with carborundum granules to help ensure even temperature of all internal components and to assist with mechanical damping of the circuitry. Six 4,700-µF, 63-V filter capacitors take up most of the rest of the space on the p.c. boards. Some film bypass capacitors for the main electrolytics and a pair of 10-watt power resistors are at the front-panel end of the p.c. boards.



Circuit Description

A complete schematic wasn't supplied with the YBA_1 I received for testing, nor was one asked for. A generalized schematic, however, appears in the owner's manual. This unit's overall topology is very much like that of many other solid-state amplifiers—with a few, and perhaps sonically significant, twists. A complementary dual-differential amp serves as the input stage. Constant-current sources for the emitter pairs are indicated but not detailed. An input coupling capacitor and a shunt feedback capacitor of some 10 µF each are housed in what appears to be an aluminum can mounted on the heat-sink near the input and output connectors. Input impedance is about 30 kilohms and is set by a base-to-ground resistor after the input coupling capacitor. Cutoff frequency for this high-pass, first-order filter should be about 0.5 Hz. However, assuming a 30-kilohm series feedback resistor, a shunt value of about 1 kilohm would be appropriate for the closed-loop gain this amp has. A 1-kilohm resistor and 10-µF capacitor form a low-frequency cutoff of 16 Hz, which doesn't jibe with the excellent square-wave response this amp exhibits at low frequencies. The 10-µF film shunt feedback capacitor is probably paralleled with an electrolytic cap housed inside the enclosure for the front-end's circuit parts.

Inverting output collectors (the NPN and PNP devices whose bases are connected to the signal input) are direct-coupled to a complementary pair of devices. These devices' emitters are referenced to the appropriate supply rail, and their collectors are tied together through a bias-spreading regulator. These complementary devices appear to be mounted without heat-sinks to the printed side of the p.c. board and are not included among the enclosed front-end components. Resistive loading of each of these last-voltage-amplifier (LVA) collectors to ground limits the open-loop gain along with emitter degeneration in the input stage. Output of the LVA is direct-coupled to a pair of complementary TO-3 drivers that are connected as emitter followers

Although overall circuitry is like that of many other solid-state amps, it has a few, perhaps sonically significant, twists.

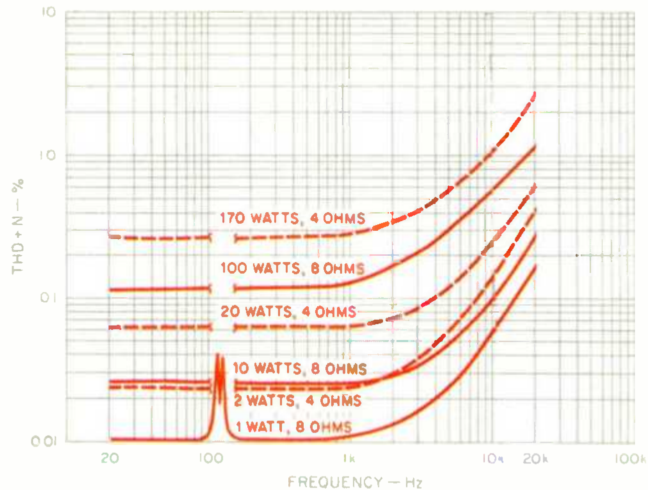


Fig. 1—THD + N vs. frequency for 8- and 4-ohm loads, each at three power levels.

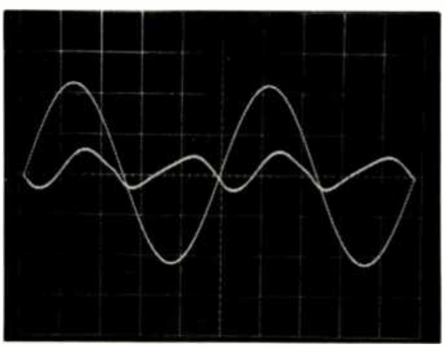


Fig. 2—Sine wave at 1 kHz, plus distortion products, for 10 watts into 8 ohms; here, THD + N is 0.029%. Note the slight kink in the distortion trace; see text. The polarity of the distortion waveform has been reversed by the analyzer.

and mounted on the main heat-sink ledge along with the output devices themselves. A resistor is tied between the emitters of the drivers to set their idling-current level. The drivers' emitters are direct-coupled to the bases of the output transistors, which consist of two NPN and two PNP TO-3 devices connected as emitter followers. The emitter resistors usually used for feedback and for increased thermal stability are not used here. Incidentally, all the devices in this circuit are bipolar transistors.

Instead of the usual insulating washers for the driver and output devices, which are usually thin pieces of mica or plastic, this unit has insulators made up of a sandwich of mica and copper sheet—i.e., mica/copper/mica. The copper pieces are bussed together and tied to the +50 V supply. The designer says this scheme's purpose is to bypass power-supply hash back to the center tap of the power transformers. (If I had been the designer, I would have connected the copper pieces directly back to the transformer center taps.) What the bussed-sandwich technique may alternatively be doing is preventing power-supply noise on the power-transistor cases from getting back to the power supply's center tap through chassis metal. Otherwise, this noise could be induced into the amplifier circuitry by capacitive and/or inductive coupling along the way. I don't really claim to understand this scheme.

No stabilizing series RC network from output to ground is used in this design, and no series RL buffer network, per se, exists. The two NPN output emitters are tied together and, along with the connection of the mating PNP output emitters, are led through two separate wires wound around the metal can surrounding the input and shunt feedback capacitors to form a coil with a small number of turns. These wires are then connected to the hot output terminal! This unusual connection is said to provide some "air feedback" and to stabilize the amp, obviating the need for the usual stabilizing network elements. André makes a point about having a minimum number of passive elements in the signal path from input to output; the devices in this path are predominantly active.

The power transformers' C-core construction may improve isolation from high-frequency noise in the power line. In such a transformer, primary and secondary coils are wound on opposite sides of a square-sided O core made up of the interleaved C pieces. Because the primary and secondary coils are not wound over or interleaved with each other, capacitive coupling between the windings is greatly reduced.

The six filter capacitors on each amp p.c. board add up to about 15,000 µF on both the positive and negative power rails of each channel—a medium to fairly high amount of capacitance in terms of energy storage for the amplifier's power rating.

The two 10-watt power resistors on the end of each amp p.c. board, mentioned earlier, are 1.5 kilohms in value. One is connected from +50 V to common; the other is connected from −50 V to common. The 30 or more mA of current flow and 3 to 4 watts per channel of power dissipation are said to pre-load the power supply and to provide some of the sonic advantages of Class-A operation without dissipating power in the output transistors themselves.

Measurements

I first ran the YBA$_1$ at one-third power (28.3 watts per channel) into 8-ohm loads for one hour. The heat-sinks got quite hot to the touch—I could only hold onto them for 2 to 3 S before having to let go. I did not run one-third power into 4-ohm loads, as I felt I might get the amp too hot and didn't

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—Lewis Lipnick, *Stereophile*, Vol. 11 No. 4, April 1988.

Recommended accessory in *Stereophile*, Vol. 12 No. 4, April 1989.

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—Ken Pohlman, *AUDIO*, November 1987.

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The YBA_1's sound was clear, transparent, and refined, with incredible definition and a wide, deep, natural soundstage.

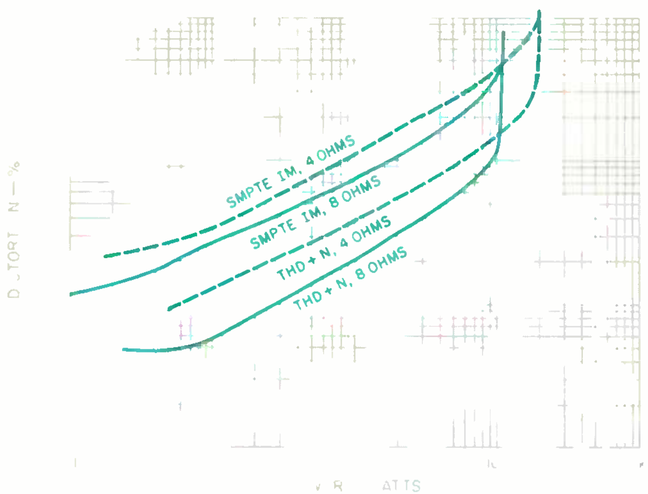

Fig. 3—SMPTE IM and THD + N vs. power for 8- and 4-ohm loads. THD + N is for a 1-kHz test signal, with distortion products measured from 400 Hz to 80 kHz.

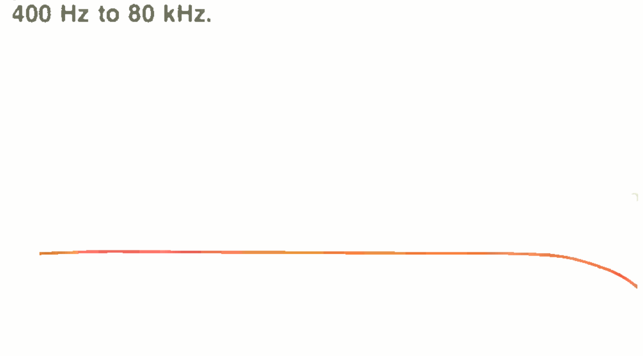

Fig. 4—Frequency response at 1 watt output into 8-ohm load.

want to risk damaging it. In all of my listening tests, this amp never got hotter than just warm.

Main power-supply voltage within the amp was ±50.5 V with 120-V a.c. line input. Voltage gain was 24.4× and 24.2× for left and right channels, or about 27.7 dB. IHF sensitivity for 1 watt into 8 ohms worked out to be 116 mV.

THD + N versus frequency, as a function of power and load, is shown in Fig. 1 for the right channel. This channel was slightly lower in distortion than the left. The lowest curve in this figure shows a nonlinearity caused by the beating of the test-signal frequency against the 120-Hz power-supply ripple. The percentage increase in distortion caused by this nonlinearity decreases as power output increases. For clarity, it has therefore been omitted from all but the 1-watt curve. I have seen this beat phenomenon in other amps in the past, and when it's present, I mention it and show it on the distortion curves. Harmonic distortion residue predominantly consists of even harmonics below 1 to 3 kHz. Above these frequencies, a distortion spike of increasing magnitude near the waveform's zero crossing causes the distortion to increase rapidly with increasing frequency. Typical harmonic distortion residue for the YBA_1 at low to medium frequencies is illustrated in Fig. 2 for 10 watts into 8 ohms at 1 kHz. Due to polarity reversal in the distortion meter's monitor output, the signal polarity of the output waveform is reversed from its actual polarity with respect to the distortion waveform. In other words, imagine the output waveform going negative in the first half-cycle displayed. The distortion glitch mentioned above is just noticeable here as a slight kink in the distortion waveform near its zero crossing in the negative-to-positive direction. Both THD + N at 1 kHz and the SMPTE-IM distortion, as functions of power and load, are plotted in Fig. 3.

Crosstalk versus frequency, with the undriven channel terminated in 1 kilohm, proved to be quite symmetrical in both directions. Results were better than −90 dB from 20 Hz to 6 kHz, decreasing to −82.5 dB at 10 kHz, 71.6 dB at 20 kHz, and 58.2 dB at 50 kHz.

Frequency response at 1 watt into 8-ohm loading appears in Fig. 4. With 4-ohm loading, the high-frequency response was within 0.1 dB of the 8-ohm response shown. Rise- and fall-times at 10 V peak to peak into 8 ohms were 3 μS. The waveform was exponential (normal) and essentially constant in rise- and fall-times from small signals up to voltage clipping. Figure 5 shows square-wave performance. The top trace is for 10 kHz into 8 ohms. The middle trace is also for 10 kHz, but into 8 ohms paralleled with a 2-μF capacitor. In the bottom trace, the test frequency is 40 Hz. Notable in these waveforms is the very low tilt at 40 Hz and the relatively low ringing, in the middle trace, for an amp without the usual output RL buffer network. All waveforms are at 10 V peak to peak.

Damping factor versus frequency is plotted in Fig. 6. Even though there is no RL output-buffer network per se, the turns of wire around the input capacitor's case form an inductor of some value and, along with a possible reduction in loop gain versus frequency, cause the output impedance to rise with frequency above 1 kHz. The reduction in damping factor (increase in output impedance) in the right channel below 100 Hz was really there in the measurement, and I have no idea what could have caused it.

IHF dynamic headroom was 1.74 and 1.53 dB for 8- and 4-ohm loading, respectively. Clipping headroom was 1.04 and 0.48 dB for 8- and 4-ohm loads, while clipping power for these conditions was 108 and 190 watts. Using the same tone-burst signal as in the other headroom tests, driving one channel into a 1-ohm load resulted in ±30 amperes of peak current—a respectable figure indeed.

IHF S/N ratio (A-weighted noise below 1 watt into 8 ohms at 1 kHz) was 99 and 98 dB for left and right channels, respectively.

A few final observations: 20-kHz clipping was clean, with no evidence of "sticking." Thermal stability, regarding quiescent current draw from the a.c. line, had some noticeable

The YBA_1 is the best solid-state amp I've heard so far. At $6,000, it's not a best buy but could well be worth it.

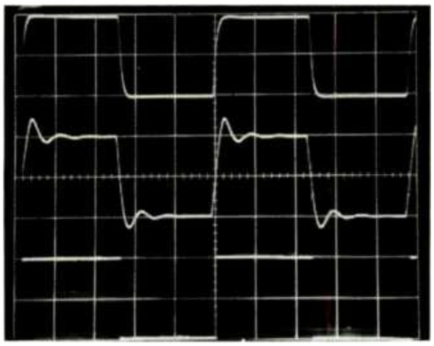

Fig. 5—Square-wave response. Top trace is 10 kHz, with 8-ohm load. With 2-μF capacitance across the 8-ohm load (middle trace), ringing is unusually low for an amplifier with no RL buffer. The tilt at 40 Hz (bottom trace) is notably low. Scales: Vertical, 5 V/div.; horizontal, 20 μS/div. for top two traces, 5 mS/div. for bottom trace.

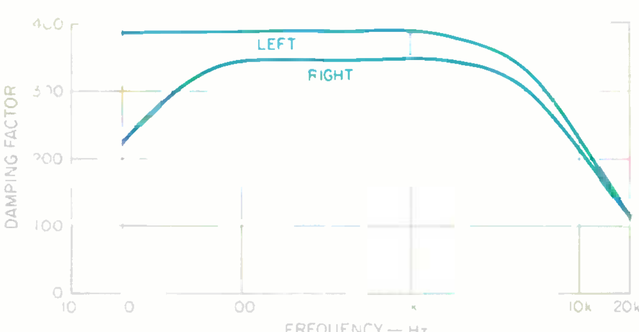

Fig. 6—Damping factor vs. frequency. Note the difference between channels; see text.

hysteresis. That is to say, if the amp was turned on and allowed to idle for a long time, current draw was about 0.4 ampere a.c. If the unit was heated up by making it produce one-third power for a while, line current could be as high as 1 ampere a.c. (or more) immediately upon cessation of the drive. Line current would then quickly drop, finally settling at about 0.6 ampere a.c. In my listening tests of the amp, I found it was predominantly cool in operation and was drawing the lower, 0.4-ampere current.

Use and Listening Tests

Equipment used to evaluate the YBA_1 included an Oracle turntable fitted with a Well Tempered arm and Koetsu Black Goldline cartridge, a California Audio Labs Tempest CD player, a Nakamichi 250 cassette deck, a Cook-King reference tube phono preamp, a Meitner PA-6i preamp, my 845 Class-A tube 100-watt mono amps, and Motif MS50 and MS100 power amps driving Siefert Research Magnum III speakers.

The YBA_1 is one of those amplifiers that sounded good to me right at the outset. I knew I had something special very quickly. André's distinctive and unusual ideas, and his attention to detail, really paid off in the sonic department.

One prong of the a.c. power plug is marked red by the factory; an interesting instruction in the owner's manual advises inserting this prong in the hot or live slot in the wall socket. With this particular amp, the marked prong was the one that would go to the longer slot, which is neutral by socket-wiring convention. Since the plug is a three-wire type, I used a three-wire to two-wire adaptor and reversed its orientation. The power-plug orientation which yielded the lowest chassis potential, when checked with a Namiki DF-100 a.c. plug-connection direction finder, agreed with the manufacturer's polarization. In use, especially when fed from the Cook-King tube phono preamp and playing records, it was easy to tell the difference between power-plug orientations. The sound was more open and natural with the plug used according to the instructions.

I would describe the sound of this amplifier—or its *lack* of sound—as very clear, transparent, and refined, with incredible definition and with very low irritation levels. Music sounded more simply "there" and real. One could unravel complex goings on with ease. Spatial replication was very good, and a wide, deep, and natural soundstage presented itself with appropriate source material. With the YBA_1 in residence, I quickly retired my 845 tube amps, as the YBA_1 sounded better. Now those 845 tubes are gathering lots of dust.

I tried the amp with IRS Beta speakers in Infinity's sound room. Although the sound with the YBA_1 wasn't quite as lush and big as it had been with the Audio Research M-300 and VTL 300W amplifiers, the YBA_1 sounded very refined, open, and spacious on the Infinity speakers.

I also tried the amp on a friend's system, one which has Apogee Acoustics Duetta Signature loudspeakers. These speakers like a lot of power, and the YBA_1 couldn't really drive them very loud. Nevertheless, within these power limitations the sound was outrageously good.

I believe that this is the best solid-state amplifier I've heard so far—certainly in my own setup. My associate, Geoff Cook, tends to agree. At $6,000 and 85 watts per channel, the YBA_1 is not everyone's best buy, but with loudspeakers of reasonable efficiency, this amp could well be worth the money. I know I am going to sorely miss the YBA_1 when I have to give it up. Thanks, Yves-Bernard André, for a very nice experience. *Bascom H. King*

RISING ABOVE THE CD JUNGLE

For most people, buying a CD player is a lot like taking a short stroll along the Amazon. And forgetting your map.

Sooner or later, you're going to get lost.

That's because the "jungle of misinformation" about CD players makes it difficult to know what's really important. And what isn't.

Take a quick look at some of the claims—digital bit structures (what are they, anyway?) ranging from 1 to 45. Oversampling rates from 2x to (quick, who's got the latest?) 16x. All this for the sake of a numbers race. And not necessarily for the sake of the music.

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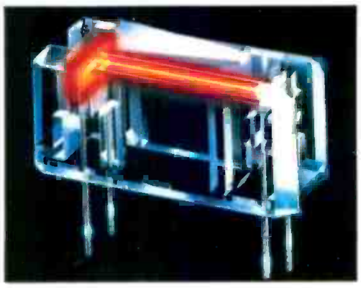

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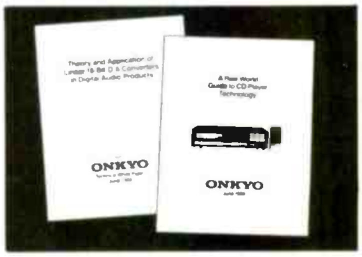

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11

ONKYO T-9090II FM TUNER

Manufacturer's Specifications

Usable Sensitivity: Mono, 10.3 dBf; stereo, 12.8 dBf.

50-dB Quieting Sensitivity: Mono, 15.8 dBf; stereo, 37.2 dBf.

S/N: Mono, 95 dB; stereo, 85 dB.

THD: Mono, 0.009%; stereo, 0.02% (both wide i.f. mode).

Frequency Response: 30 Hz to 15 kHz, +0.5, −1.0 dB.

Alternate-Channel Selectivity: 95 dB (super-narrow mode).

Adjacent-Channel Selectivity: 45 dB (super-narrow mode).

Capture Ratio: 1.0 dB.

Image Rejection: 100 dB.

I.f. Rejection: 100 dB.

AM Suppression: 60 dB.

Separation: 55 dB at 1 kHz, 33 dB from 70 Hz to 10 kHz (both wide mode).

Output Level: 0 to 1.5 V.

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

Dimensions: $18\frac{5}{16}$ in. W × $4\frac{1}{16}$ in. H × $15\frac{1}{4}$ in. D (46.5 cm × 10.3 cm × 38.7 cm).

Weight: 18.7 lbs. (8.5 kg).

Price: $749.95, including wood side panels.

Company Address: 200 Williams Dr., Ramsey, N.J. 07446.

(Originally published July 1988)

No, this is *not* a replay of an earlier review. When I evaluated the Onkyo T-9090 back in 1985, I gave it high marks and assumed that its performance was about as good as current technology would permit. Well, advances in technology never end, and that remarkable tuner's successor, the T-9090II, represents so much more than a cosmetic face lift that I felt it was worth evaluating and measuring. Most of the features on the original version have been retained, and additional features have been incorporated for convenience and compatibility with FM cable systems.

The T-9090II FM-only tuner has substantially improved selectivity over the previous model, owing to entirely new i.f. filters. It also has a more sophisticated Automatic Precision Reception (APR) system that allows you to program preferred reception modes along with station presets. The modes selected by the APR system include r.f. stage gain (distant/local), i.f. bandwidth (wide, narrow, or super narrow), stereo/mono, high blend, and antenna A or B. Although the APR system works automatically to provide the best field strength, lowest distortion, and lowest noise for each captured station, the above-named modes can be overridden if you wish.

Two antennas can be connected to this tuner. These may be two conventional FM antennas or one FM antenna and one cable FM connection. During tuning, each antenna input is sampled continuously, and the better signal is automatically chosen. This type of "diversity" reception can yield clear local signals from antenna A, for example, while antenna B can be oriented toward a more distant city that may provide weaker (but still usable) signals. Alternatively, antenna A might be used for over-the-air FM reception, and antenna B could be connected to a cable television feed for simulcast FM soundtracks of stereo pay-TV movies or music video channels.

The remote supplied with the T-9090II allows complete control over most front-panel functions. It can even drive the motorized potentiometer that controls level at the variable output jacks. Preset stations can be sampled for 5 S each, and timer preset tuning (with an external timer) can be used to select preset stations in a predetermined order for unattended recording.

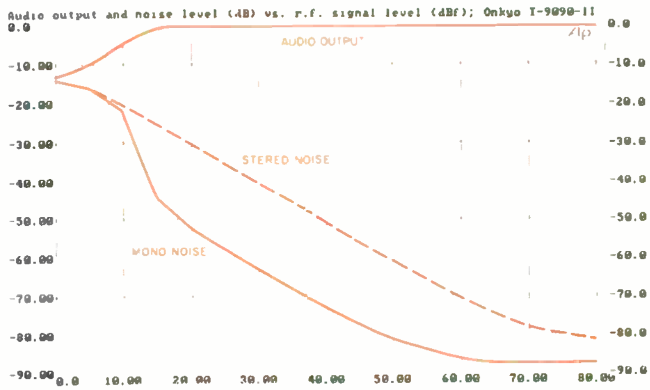
The T-9090II tunes in 50-kHz steps during automatic scanning, but manual adjustment is also possible, via the front panel, in 25-kHz steps. A multi-function fluorescent display shows station frequency, in MHz, to three decimal places. A linear scale shows relative r.f. input signal strength in 10-dB steps, and a numeric indicator also can show r.f. strength in dBf. The same numerical indicator can be set to show the signal threshold for scan tuning.

Control Layout

The on/off pushbutton is at the lower left of the front panel. To its right, along the lower edge of the panel, are a "Shift" button and 10 numbered buttons for setting or recalling up to 20 preset station frequencies (numbered 1 to 10 or, in combination with the "Shift" button, 11 to 20). Just above these are eight smaller keys for setting the scanning threshold (17, 27, or 37 dBf), displaying signal strength in dBf, initiating preset scanning, switching between manual and automatic modes, tuning up or down, storing memory presets, and calling up "Auto Memory" mode. The latter automatically finds stations which can be received satisfactorily. It then stores them in order of ascending frequency, starting from the currently tuned frequency and whatever memory position you select.

On the right half of the display panel, a number of indicator lights show the various tuning modes selected manually or by the APR system, as well as muting status, stereo reception, and center-tuned status. Below them are 20 small, numbered indicator lights for the station presets.

Nine additional pushbuttons and one knob are at the right end of the panel. The five buttons in the top two rows select local or distant receptions, the i.f. bandwidth, high blend, stereo or mono, and antenna A or B. The remaining buttons control FM muting, restore APR operation after any of its settings have been overriden, indicate the next station to be selected during external-timer operation, and switch timer-controlled station selection on and off. The small rotary knob controls the output level for the variable output jacks on the rear panel.



A

B

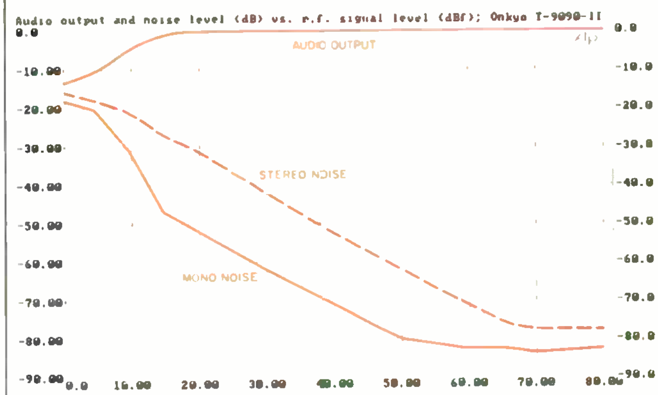


Fig. 1—Mono and stereo quieting characteristics for wide (A) and narrow (B) i.f. modes.

Onkyo's T-9090II tuner automatically selects whichever of its two antenna inputs has the better signal.

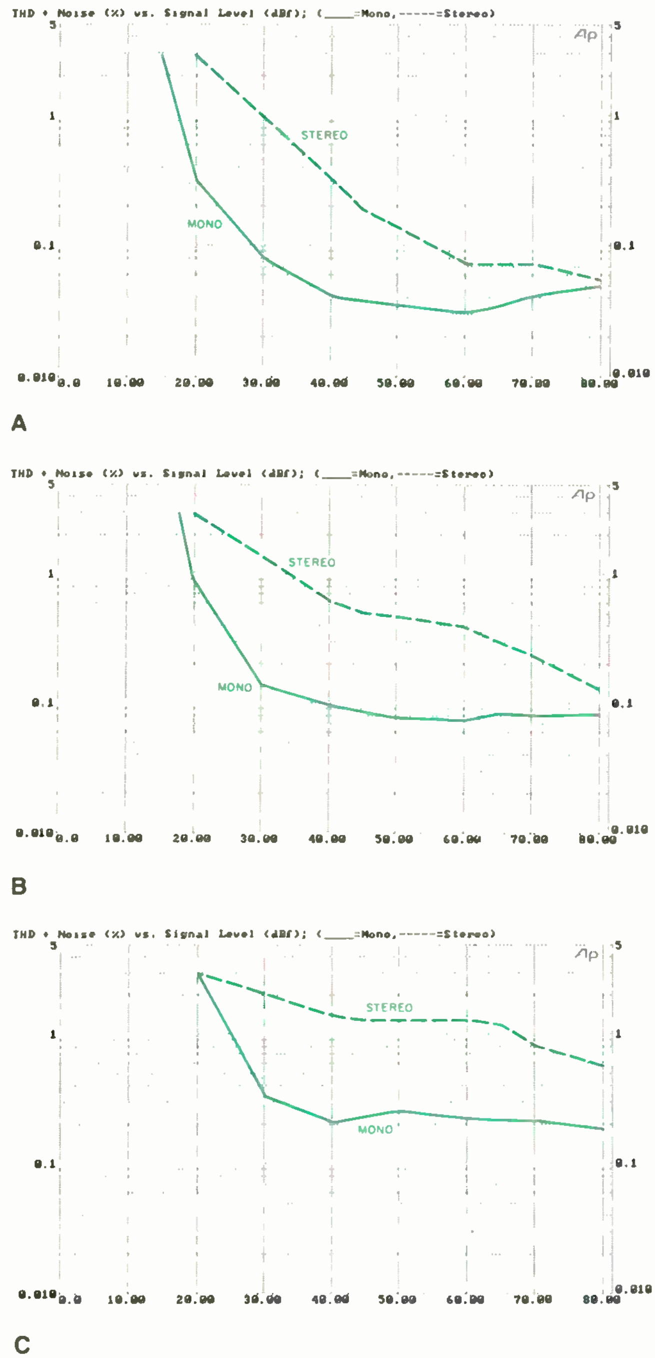


Fig. 2—THD + N vs. signal strength at 1 kHz for wide (A), narrow (B), and super-narrow (C) i.f. modes.

In addition to the two coaxial (75-ohm) antenna inputs and the fixed- and variable-level output jacks on the rear panel, there are two more jacks, labelled "V" and "H." These jacks can be connected to any oscilloscope's vertical and horizontal inputs for observing and minimizing multipath problems by antenna reorientation. For those who use flat, 300-ohm, twin-lead transmission lines from their antennas, Onkyo supplies 300/75-ohm matching transformers.

Measurements

In testing this tuner's quieting characteristics (Figs. 1A and 1B), I measured 12.0 dBf of mono usable sensitivity in the wide i.f. mode; this increased slightly, to 11.5 dBf, in the narrow mode. Stereo usable sensitivity was 20 dBf in the wide mode and 18 dBf in the narrow mode. Because no substantial difference in quieting characteristics occurred between the narrow and the super-narrow modes, no graph is shown for the latter. In mono, 50-dB quieting was achieved with 18 dBf of signal input in the wide i.f. mode; the same level of quieting was reached in the narrow mode with only 16 dBf of input signal. In stereo, the 50-dB quieting point was reached with a signal level of 39 dBf in the wide i.f. mode and 35 dBf in the narrow mode.

Best signal-to-noise ratio that I was able to read was 87 dB in mono (at 65 dBf) and 81 dB in stereo (at 80 dBf), using the wide i.f. mode. Outstanding though they are, these figures fall somewhat short of Onkyo's claims. I suspect that they are limited by the residual noise inherent in my FM generator. (I have never been able to measure better than 88 dB or so with this instrument.)

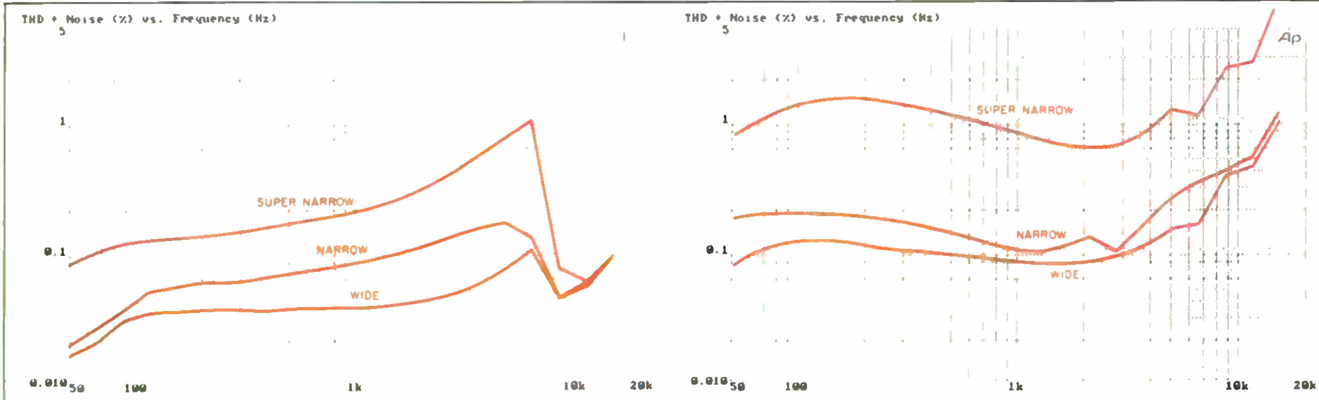
Figures 2A, 2B, and 2C show THD + N versus signal strength. Lowest distortion for a 1-kHz modulating signal was, of course, obtained with the wide i.f. mode (Fig. 2A). Under these conditions, THD + N was only 0.035% in mono (for a 65-dBf signal) and 0.056% in stereo (for an 80-dBf signal). At 65 dBf, stereo THD measured 0.075%. The narrow mode yielded readings that were still below 0.1% in mono and just over 0.1% in stereo for strong, 80-dBf signals (Fig. 2B). Under conditions of extreme interference from adjacent channels, the super-narrow setting might be justified. This is true even though THD + N (for a 65-dBf signal) was more than 0.2% in mono and reached the 1% level in stereo in this mode (Fig. 2C).

Plots of THD + N versus frequency, for a strong r.f. input signal, are shown in Fig. 3A (mono) and Fig. 3B (stereo). In each case, the bottom curve represents results obtained using the wide i.f. mode, the middle curve using the narrow mode, and the top curve using the super-narrow mode.

The audio generator output of my Audio Precision test gear can be equalized using a wide variety of curves. To measure frequency response of the Onkyo T-9090II, I simply used a 75-µS *pre-emphasis* curve on the modulating audio signals. (This is the reciprocal of the de-emphasis characteristic required in FM tuners sold in the U.S.). In the response plot (Fig. 4), I have expanded the vertical scale to 2 dB per division. The response curve shows a superbly flat characteristic from 20 Hz to 10 kHz with a roll-off of a bit more than 1.0 dB at 15 kHz.

Separation versus frequency (Fig. 5) is plotted for all three i.f. bandwidths. At 1 kHz, separation ranged from over 50

I couldn't second-guess this tuner. It picked the best mode settings for every station I tried.



A B

Fig. 3—THD + N vs. modulating frequency for mono (A) and stereo (B).

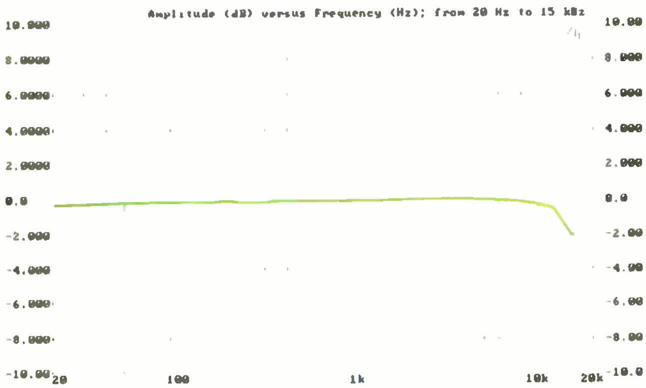


Fig. 4—Frequency response. Note vertical scale of 2 dB/div.

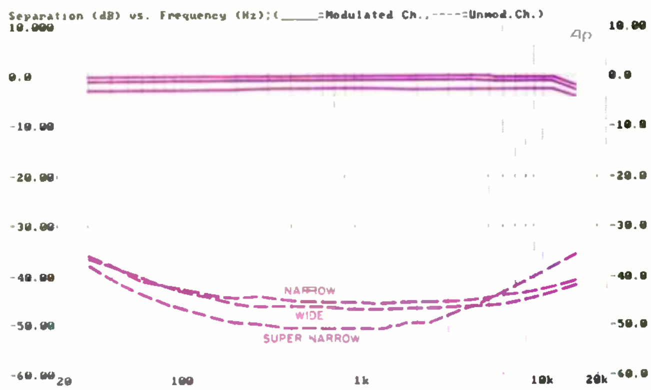


Fig. 5—FM frequency response (top curves) and separation vs. frequency (bottom curves) for all three i.f. modes.

dB to 45 dB, depending on the bandwidth setting. The solid curves represent the output of the modulated channel, and the dashed curves show the output of the unmodulated channel. Surprisingly, the *middle* dashed curve was obtained with the tuner set to the wide i.f. mode. The top dashed curve was plotted using the narrow mode, and the bottom dashed curve was made with the super-narrow mode. It's interesting to note that, at mid-frequencies, the best separation was obtained with the tuner set to where one would expect the worst separation! The effect of the super-narrow bandwidth is realized at higher frequencies, but even then, separation at 10 kHz remained nearly 40 dB in this mode!

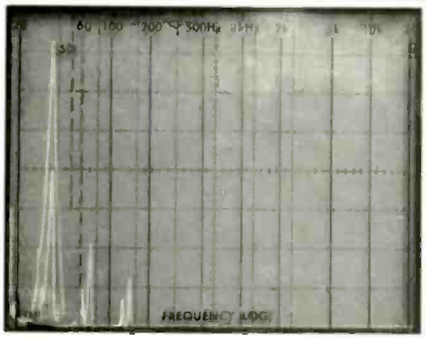
The super-narrow i.f. setting, while minimally affecting stereo separation, does have a rather marked effect on crosstalk and distortion, as shown in Figs. 6A and 6B. In these spectrum analyses, the sweep is linear, from 0 Hz to 50 kHz. In Fig. 6A (wide mode), you can see that the crosstalk products to the right of the main 5-kHz signal are much smaller in amplitude and more free of modulation noise than they are in Fig. 6B (super-narrow mode).

I measured a capture ratio of 1.0 dB in the narrow mode and 1.5 dB in the wide mode. Image, i.f., and spurious rejection were all better than 100 dB. In the super-narrow mode, alternate-channel selectivity was an outstanding 95 dB and adjacent-channel selectivity was greater than 40 dB. In the wide mode, alternate-channel selectivity measured 38 dB. AM suppression was 62 dB, better than the manufacturer's claim.

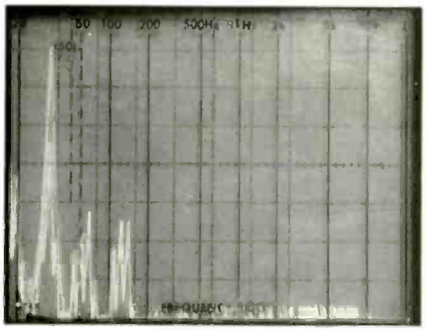
Use and Listening Tests

I tried, repeatedly, to second-guess the APR system built into this "intelligent" unit, but with no success. I tuned to several stations of varying quality and manually tried to set

For all its improvements, the T-9090II costs little more than Onkyo's earlier version, despite the rise of the yen vs. the dollar.



A



B

Fig. 6—Separation and crosstalk components for a 5-kHz modulating signal, plus subcarrier and sideband components, for wide (A) and super-narrow (B) i.f. modes.

up listening conditions that were contrary, in one way or another, to what I thought the APR system would have chosen. Each time I pressed the APR button, the system selected a set of operating conditions that yielded better results than those delivered by the conditions that I had chosen. In short, the APR system works—you can trust it completely.

To check out the two-antenna diversity reception capability in my metropolitan New York area, I hooked up a multi-element outdoor antenna to one antenna input and an indoor dipole to the other. I was able to rotate the outdoor antenna to favor some distant Connecticut stations, while the indoor dipole was oriented toward the major, powerful stations in my area. Sure enough, when I set the tuner to automatic scanning, it switched back and forth between the two antennas exactly when it should have, and my total station count reached an astounding 58! There's no way I could have picked up this many usable signals from a single antenna, unless I rotated the antenna as each new signal was tuned in. Even at that, I probably would have missed some, since a single antenna might well have been rotated so far "off course" for certain signals that the scanning threshold would have overlooked them completely.

If you check out the T-9090II for yourself, don't be put off by its elaborate display area. True, there are an inordinately high number of indicators, but in this case, they are not there just for ornamentation. Each light really provides you with useful—perhaps even essential—information about the status and mode of the tuner's operation. If you are as much of an FM fan as I am, and if you are blessed with a couple of FM stations that really care about the signals they transmit, this is definitely a tuner worth considering.

On a practical level, I am amazed to see that the price of the T-9090II, for all its enhanced features and higher performance, is not much more than that of the earlier Model T-9090—and this in the face of the steadily rising value of the Japanese yen against the dollar. Obviously, Onkyo's talents apply not only to FM design engineering, but to production efficiency as well. *Leonard Feldman*

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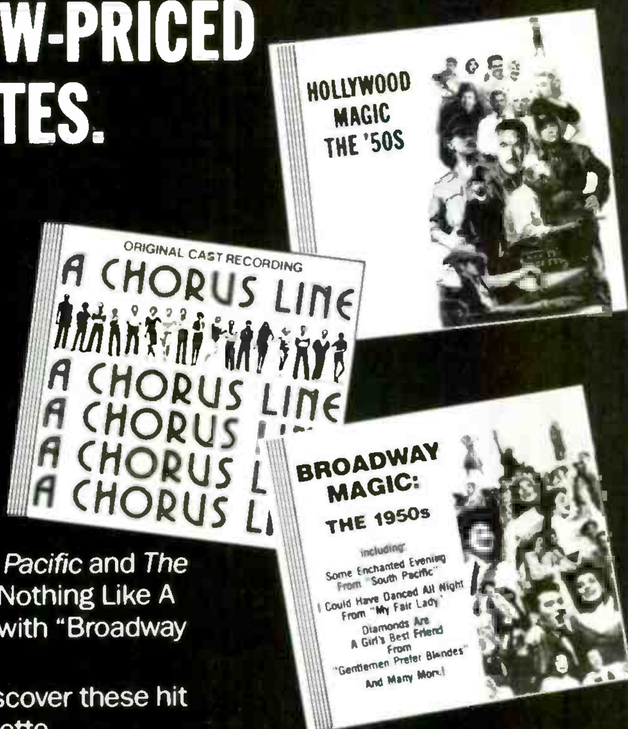

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"Finian's Rainbow"
"Pal Joey"
"Bells Are Ringing"
"Bye, Bye Birdie"
"Anyone Can Whistle"
"On The Town"

12

REVOX B260-S FM TUNER

Manufacturer's Specifications

Usable Sensitivity: 10.8 dBf.

50-dB Quieting Sensitivity: Single-r.f. mode, 13.2 dBf in mono and 34.8 dBf in stereo; double-r.f. mode, 16.7 dBf in mono and 38.3 dBf in stereo.

Selectivity: Wide, 50 dB; narrow, 100 dB.

Frequency Response: 20 Hz to 15 kHz, ±0.5 dB.

S/N: Greater than 80 dB.

THD: Mono, less than 0.07% at 1 kHz.

Muting Threshold: 2.0 µV.

Stereo Threshold: 10 µV.

Auto Tuning Threshold: Local, 100 µV; distant, 4 µV.

Separation: 43 dB at 1 kHz; 15 dB in Blend 1 and 7 dB in Blend 2.

Calibration Tone: 400 Hz at 40 kHz equivalent deviation.

Power Requirements: 120 V a.c., 50/60 Hz; 20 watts, 5 watts standby.

Dimensions: 17.7 in. W × 4.3 in. H × 13 in. D (45 cm × 10.9 cm × 33.2 cm).

Weight: 15.4 lbs. (7 kg).

Price: Tuner, $2,500; B208 remote control, $160.

Company Address: 1425 Elm Hill Pike, Nashville, Tenn. 37210.

(Originally published September 1988)

There aren't too many over-$2,500 FM tuners out there anymore. I'm told that the old Sequerra tuners, which sold for that much and more, are still being made, but the Marantz 10B that started it all is not. So when I came across one (old FM buff that I am), my first inclination was to try to determine if the price was justified. After all, FM reception quality is ultimately limited by FCC broadcast standards set more than 40 years ago and amended for stereo in 1961. You can't, for instance, receive audio frequencies above 15 kHz, and usable sensitivity is limited by the thermal noise floor of the antenna input impedance. So what makes a tuner like the Revox B260-S worth $2,500? Plenty—but most of it has to do with convenience features and the tuner's ability to tailor reception specifically to the environment and signal conditions prevailing in your location.

Let's begin with some of the more innovative features. You can preset up to *60* stations. That may well be more than the number of presets that any other tuner can memorize, but as I see it, the real benefit is the ability to divide those memorized station frequencies into subgroups. Suppose you like jazz and classical but your spouse or roommate prefers soft rock and theater musicals. You could find all the stations that transmit your kind of music and store them under one subgroup heading, while the preferred stations of other household members could be stored under other subgroup headings. As many as nine subgroups can be organized and stored on the B260. Of course, all of the more conventional tuning modes are available, such as scanning (with brief auditioning as each usable signal is encountered), direct call-up of memorized stations, scanning within subgroups, and manual tuning. Another interesting feature is called "Auto Tune." It is a one-touch, initial setup programming system in which you need only tune in a station, touch one button, and the next available preset location is automatically assigned to the new station.

Besides station frequencies, the preset memories hold whatever settings you use to tailor the tuner's operation to your specific needs and preferences, so you don't have to go through all those switches and adjustments every time you call up the stations. Even the tuner's output level can be set separately for each station, so you can use the same volume setting whether you're listening to a quiet classical station or one of those stations which tries to be loudest on the dial by modulating up to (or beyond) the legal limits. Information can also be programmed about which of the B260's two antenna inputs should be selected for each preset station, as can the degree of stereo blend, i.f. bandwidth, and even r.f. sensitivity. There are two r.f. gain settings about 4 dB apart—just enough to make overly powerful stations, which might otherwise overload the front-end, usable again.

The B260 display area has provisions for entering the call letters associated with each of the station frequencies you memorize. Up to four characters can be memorized for each station. I haven't seen that on a tuner since Larry Schotz first introduced it on his memorable Micro/CPU 100 tuner design more than a decade ago, though some recent receivers also offer it.

Most home tuners let you switch to mono if stereo background noise or multipath interference becomes unbearable for a given signal, and some tuners feature a blend position which reduces noise at the expense of some stereo separation. The B260 offers *two* blend settings. One maintains about 15 dB of separation across the entire audio spectrum (not just reduced separation at high audio frequencies). The other provides only minimal separation (about 7 dB) for those situations in which most tuners would have to be switched fully to monophonic mode to make the background noise level tolerable.

The B260's control panel could be used to teach a course in human engineering.

To assist the user in setting up proper audio output levels for recording, there's a built-in calibration tone. As with most tuners having this feature, the tone's output is at the average level of most reasonable FM stations, equivalent to 40-kHz signal deviation.

It goes without saying that, while these convenience features are all well and good, they alone wouldn't provide sufficient justification for the high price of this FM tuner—unless its sensitivity, selectivity, and other measured parameters were among the best available. Happily, they are, as you will see from my lab measurements and listening tests.

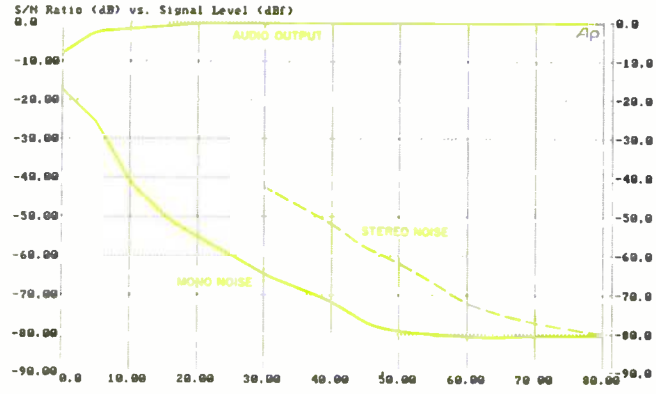
Control Layout

If I am ever asked to teach a course on industrial design and good human engineering, I think I will borrow a Revox B260 to illustrate how a tuner's front panel *should* be designed. To begin with, the B260's low-profile design matches that of Revox's recently introduced B250 integrated amplifier. Both units are about 1½ inches lower in height than earlier Revox audio components and are available in black with gold trim instead of the company's traditional gray and silver finish.

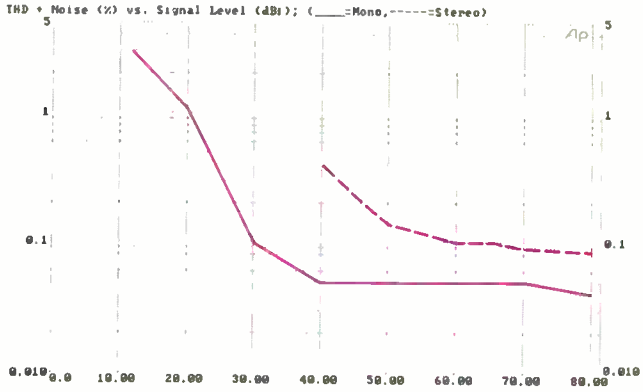
The B260's programming controls are placed behind a hinged, smoked-glass cover, completely separated from the main operating controls. There is little chance of a child (or even you) accidentally upsetting previously programmed settings.

When the power switch at the extreme right of the panel is pushed, the tuner is turned on and the last received station is reactivated. Two large rocker pads just to the left are used for changing stations; their functions and illuminated labels

The B260 stores not only frequencies but call letters, station formats, and tuner settings for up to 60 stations.

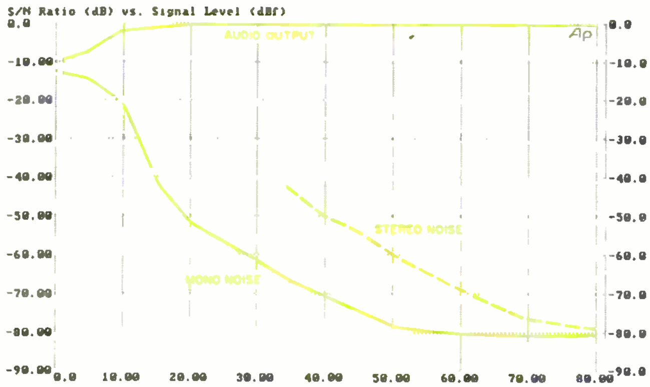


A



A

B



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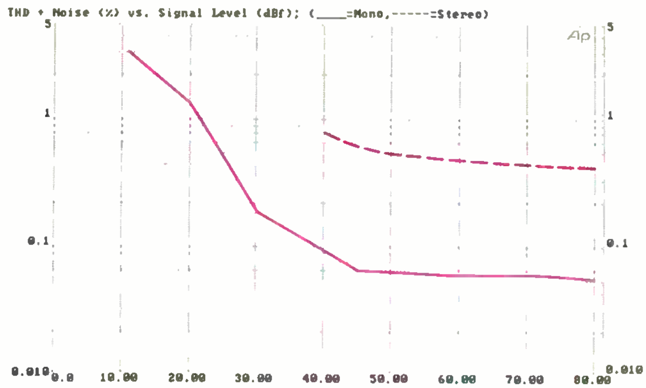


Fig. 1—Mono and stereo quieting characteristics, wide i.f. mode, for single-r.f. (A) and double-r.f. (B) settings.

Fig. 2—THD + N vs. signal strength at 1 kHz for wide (A) and narrow (B) i.f. modes.

change according to the setting of two buttons ("Tuning" and "Station") behind the smoked-glass panel.

In "Station" mode, the two rockers scan the preset memories. The right-hand rocker (now "P-Type Scan") scans up or down through the station subgroup, while the left-hand rocker ("Station Scan") scans up or down through all 60 programmed station memories, regardless of subgroup.

When the B260 is in "Tuning" mode, the right-hand rocker's function becomes "Autotuning," activating ascending or descending automatic station scan. The other rocker functions as "Frequency Step," and it is used for manual tuning in steps of 10 or 50 kHz, as determined by one of the secondary controls. That 10-kHz step feature is extremely useful if you suffer from adjacent-channel interference, since it enables you to tune slightly away from the interfering sidebands of the adjacent-channel transmission without detuning enough to introduce serious distortion.

Small pushbuttons labelled "P-Type" and "Enter" set up the mode for entering a subgroup identification and activate the input function after retrieval or programming of a station memory or a program subgroup. Ten numerical keys are arranged across the upper left of the panel, along with a larger rocker labelled "Display." Pushing this pad changes part of the display area below, which can be cycled through to read station call letters, tuned-to frequency, or both. (Station memory number and program subgroup number are always displayed.) A stereo indicator light is also part of this display, which occupies nearly the entire lower-left quarter of the panel.

The secondary controls referred to earlier are under the smoked-glass cover, which swings smoothly down and out of the way after a pushbutton labelled "Open" is depressed. At the same time, a small secondary display, only dimly visible when the cover is up, becomes clearly visible amidst an array of secondary pushbuttons. A "Step" button selects 10- or 50-kHz frequency increments when you are in the manual tuning mode, which is selected by pressing the nearby "Tuning" button. Pressing the "Station" button, or closing the hinged lid, cancels the manual tuning mode. The "Recall" button restores the previously tuned frequency. A

This tuner offers some unusual choices, such as two levels of stereo blend and r.f. gain.

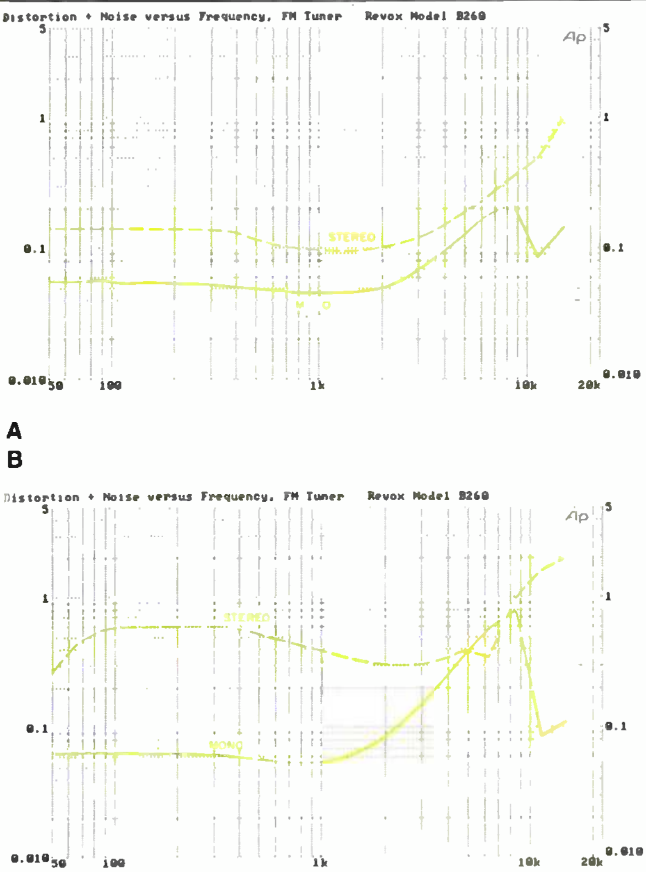


Fig. 3—THD + N vs. modulating frequency for wide (A) and narrow (B) i.f. modes.

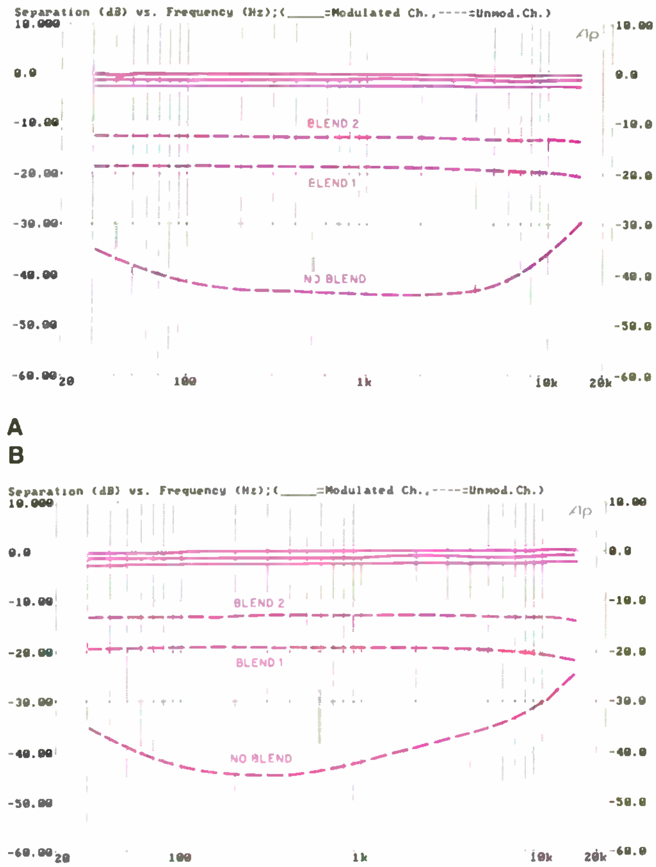


Fig. 4—FM frequency response (solid curves) and separation vs. frequency (dashed curves) for wide (A) and narrow (B) i.f. modes. Note the two blend levels. Solid curves (top to bottom) are without blend, with Blend 1, and with Blend 2.

button labelled "Search" determines the signal-strength threshold that will apply during automatic tuning or scanning. Other small buttons select the antenna input desired, r.f. gain selection (local or distant), i.f. bandwidth, mono or stereo reception, stereo blend levels, muting, and such additional functions as are required for the sophisticated programming modes featured in this tuner. Two of the remaining buttons are used to access alphanumeric characters for setting up station call letters or whatever label you choose to program in. The small display, which shows up in this area when the hinged cover is lowered, is multifunctional. Among its parameters are signal strength, center tuning, muting, antenna choice, mono mode, high-blend mode, search mode, audio level, i.f. bandwidth, and r.f. mode.

The rear panel houses the two antenna inputs, stereo output jacks, and a pair of jacks intended for connection to an oscilloscope for checking multipath during antenna orientation. There's also a multipin connector for use with Revox's multiroom infrared control receiver or their programmable controller when integrating the tuner with other Revox audio components. When the tuner is in "Station" (automatic tuning) mode, an optional wireless remote, the Revox B208, can control all of the main functions of the B260 except "Display."

Measurements

Both of the B260's antenna inputs have impedances of 75 ohms, but neither uses the "F-type" connector we are accustomed to. Revox supplied an appropriate adaptor (to which a coaxial cable could be wired) as well as a "splitter," in case you want to connect your TV antenna or cable feed to the tuner as well. I used the adaptor to connect my 50-ohm generator output to the 75-ohm inputs, and I took that slight impedance mismatch into account in those measurements where it would significantly affect the results. (Normally, I use my generator's 300-ohm output, but the B260 has no 300-ohm input, and transformers would cause excessive signal losses.)

Figures 1A and 1B are graphs of S/N ratio versus input signal level for the single-r.f. (distant) and double-r.f. (local)

There's no subcarrier leakage, so you can tape from the B260 without using your deck's MPX filter and cutting the highs.

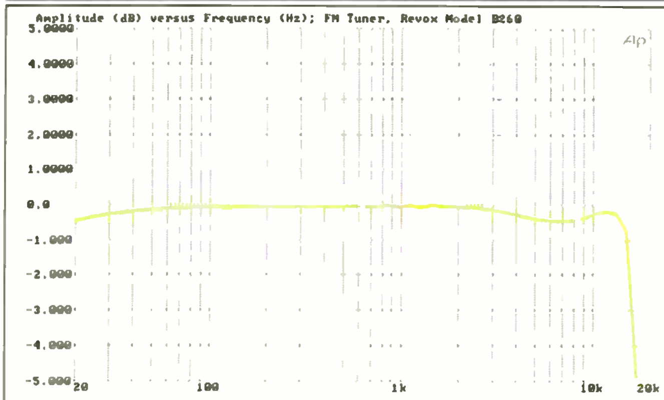
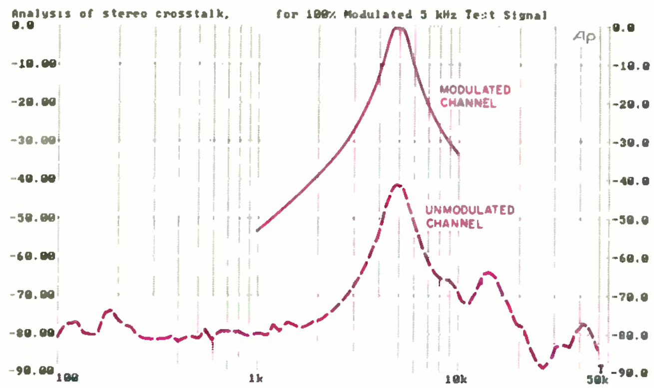


Fig. 5—Frequency response. Note vertical scale of 1 dB/div.



A

B

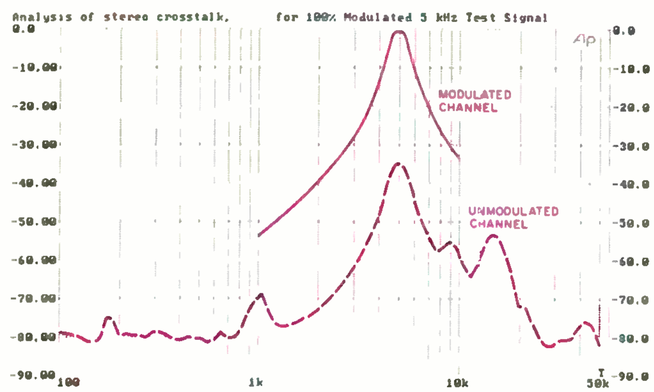


Fig. 6—Separation and crosstalk components for a 5-kHz modulating signal, plus subcarrier and sideband components, for wide (A) and narrow (B) i.f. modes.

modes, respectively. (Since there was virtually no difference in quieting characteristics between the wide and narrow i.f. modes, I did not deem it necessary to re-plot these graphs for the narrow mode.) The chief differences between the results of Figs. 1A and 1B can be seen at low input-signal levels. For the distant mode, 50-dB quieting in mono was achieved with an input signal of 14 dBf, whereas with the double-r.f. setting, it took nearly 20 dBf of signal strength to reach the same −50 dB noise level. The difference was less profound in stereo, since stereo quieting inherently requires far more input signal to reach a noise level of − 50 dB. In this case, it took exactly 38.0 dBf to reach that level of background noise using the distant setting, and 40 dBf in the local mode.

Figures 2A and 2B show how THD + N varied with signal strength. (For this and all further tests, I used the distant mode.) From these graphs, you can see that the 3% point (usable sensitivity) was reached at a signal level of between 11.0 and 13.0 dBf, depending on whether the narrow or wide i.f. mode was employed. On my sample, the stereo switching threshold was set higher than specified, so I could not measure performance in stereo at signal levels below 30 dBf (for S/N measurements) or 40 dBf (for THD + N measurements). At strong signal levels, where THD + N is primarily THD, distortion for a 1-kHz signal measured 0.05% in mono for the wide i.f. mode and 0.055% for the narrow mode. In stereo, THD reached a low point of about 0.09% in the wide mode and about 0.4% in the narrow mode.

Figures 3A and 3B show how THD + N varies with frequency. (The signal level used for these tests was 65 dBf.) Figure 3A gives results for the wide i.f. mode, while Fig. 3B illustrates the somewhat higher THD + N levels that result when the tuner is operated in the narrow mode.

Figures 4A and 4B show frequency response of a modulated stereo channel versus separation, measured at the output of the opposite, unmodulated stereo channel. (The dashed curves represent separation, measured in dB.) Figure 4A shows results obtained in the wide i.f. mode; Fig. 4B, the narrow i.f. mode. With either of the blend settings, separation remained virtually the same regardless of whether the tuner was set to the wide or narrow i.f. mode. Note, too, that the output from the modulated channel decreased a bit as I switched from no blend to Blend 1 and then to Blend 2. In measuring separation, therefore, I had to reference each separation curve to the appropriate solid-line curve with which it was associated. Thus, while separation in the Blend 1 setting appears to be about 19 dB, in reality it was more like 16 dB when referenced to its corresponding, modulated-output reference level (the lowest of the three solid curves). Similarly, separation in Blend 2 was actually about 11.5 dB rather than what appears to be 13 dB.

In the no-blend mode, maximum separation at 1 kHz in the wide i.f. mode (Fig. 4A) measured 43.5 dB at 1 kHz, 41.2 dB at 100 Hz, and 36 dB at 10 kHz. Switching to the narrow i.f. mode (Fig. 4B), maximum separation decreased to 41.6 dB at 1 kHz and 30 dB at 10 kHz. Although the solid curves of Figs. 4A and 4B gave me a good idea of overall frequency response of the tuner, the vertical scale used in those graphs is so compressed that almost any tuner's response would look perfectly flat. I therefore reset my Audio Preci-


klipsch

The switchable r.f. gain helped make strong stations more listenable, and the blend circuits did the same for weak ones.

sion test system to expand the vertical scale from 10 to 1 dB per division, and replotted the same measured frequency data; the results are shown in Fig. 5. Here you can see that the response does roll off just a bit at the low end (about −0.2 dB at 30 Hz). It also dips a bit at the high end (about −0.4 dB between 6 and 8 kHz) and rises again above 10 kHz before finally rolling off at around 15 kHz. Response was down about 0.7 dB at 15 kHz.

Since getting this new test gear, I have gradually abandoned my practice of photographing the screen of my spectrum analyzer 'scope to show separation and crosstalk for a 5-kHz signal modulating just one channel in the stereo mode. Now I can plot this information on an easy-to-read grid, as shown by Figs. 6A and 6B (wide and narrow i.f. modes, respectively). I believe that, despite the fact that the sweep is made using a somewhat wider bandpass filter than was used on my spectrum analyzer, results are clearer and easier to interpret. The plot is logarithmic and extends from 100 Hz to 50 kHz. The upper trace represents the desired 5-kHz output from the modulated channel. Its peak is set precisely at 0-dB amplitude, making it much easier to read the separation, which turned out to be just over 40 dB in the wide i.f. mode and almost exactly 35 dB in the narrow mode. More important, it is easy to see that, in the narrow i.f. mode (Fig. 6B), not only did the levels of second and third harmonic components (10 and 15 kHz) increase, but a spurious component can be seen at around 1 kHz. No such component is evident in the wide i.f. mode (Fig. 6A).

Capture ratio was highest in the narrow i.f. mode, measuring just under 1.0 dB. Alternate-channel selectivity measured 53 dB in the wide mode and greater than 100 dB in the narrow mode. Spurious-response and i.f. rejection also exceeded 100 dB, while AM rejection measured more than 60 dB. There was absolutely no evidence of spurious subcarrier frequency output when the tuner was operating in the stereo mode and input signals were in stereo. If such products were there, they were below the residual noise levels and, therefore, I could not read them. If you want to record FM programs from this tuner onto your cassette deck, you'll never have to sacrifice a few kHz of response by switching in your deck's MPX filter.

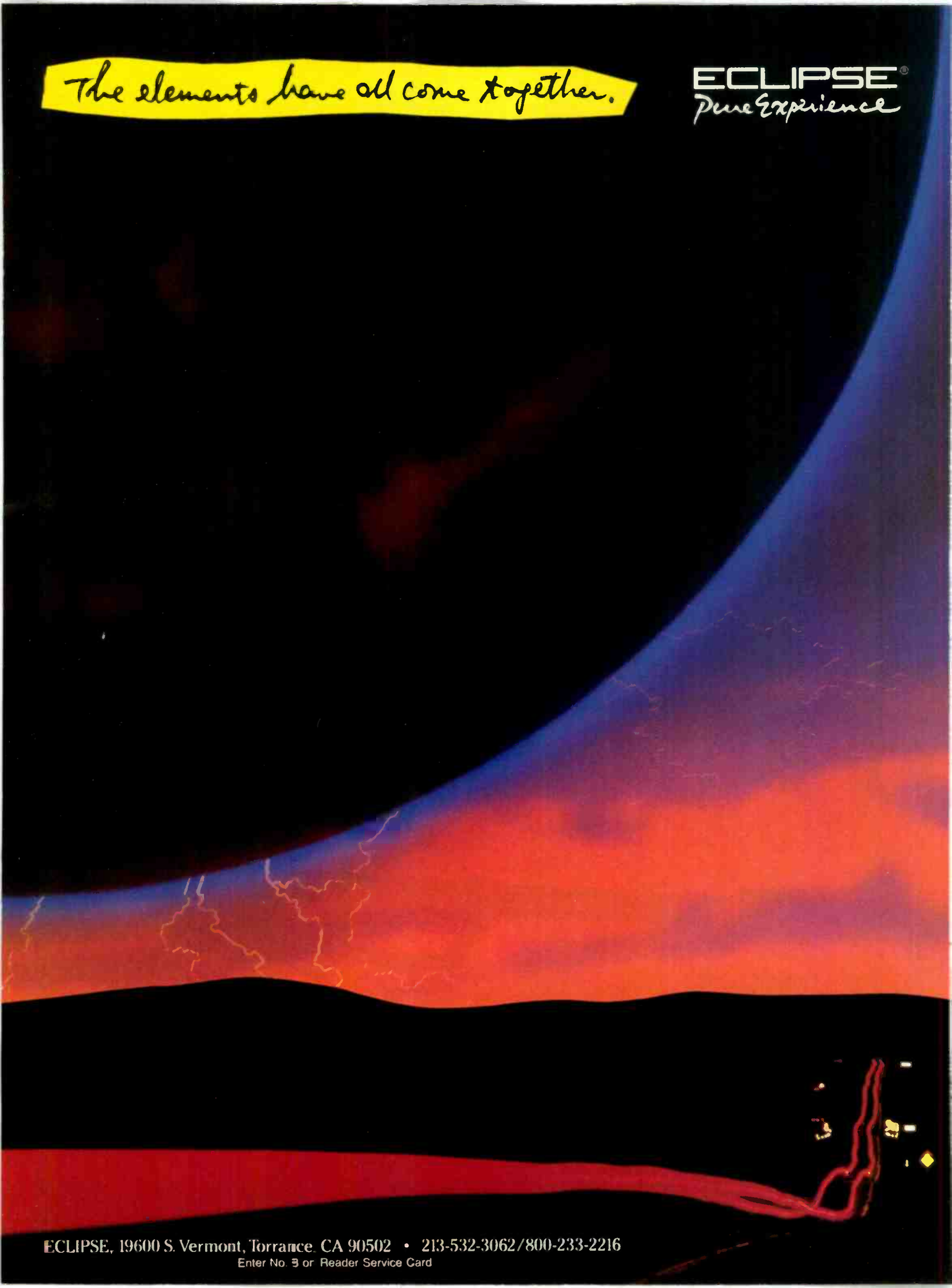
Use and Listening Tests

I began to fully appreciate the features of this remarkable tuner only when I started listening to some of my favorite stations. The classical music station I most enjoy is noted for its practice of keeping modulation levels very low to allow for sudden dynamic peaks, especially when playing CDs, without having to ride gain or use compressors excessively. But that has always meant that, when I switched from that station to my next most favorite stations (another classical one and a station specializing in jazz and early pop and rock), I usually had to run for the volume control on my preamplifier. Not with the Revox B260! I simply set the audio output level about 4 or 5 dB higher for my favorite low-modulating station. Now as I switch back and forth between stations, they all sound about equally loud.

In absolute terms, the Revox B260 was not the most sensitive tuner I've ever measured, but that didn't seem to reduce the number of usable signals I was able to pick up. In fact, I was even able to get a couple of local stations that I have traditionally not bothered to count in tuner reports because they usually caused unacceptable levels of r.f. intermodulation distortion. In the case of the Revox B260, I simply switched to the double-r.f. setting (I wish they had called the two settings "Local" and "Distant" since that's what they really are, functionally) and reduced the r.f. gain enough to solve the problem. As for weak-signal stereo reception, I was really surprised to find how effective stereo can be with only a bit more than 10 dB of separation, providing that separation is uniform at all audio frequencies, as it is in the case of the Blend 2 setting. Trading off some separation in return for background noise that's almost as low as that heard in mono is worthwhile.

After completing all of my bench and listening tests, I managed to obtain Revox's separately sold B208 remote control. This multipurpose remote operates many other Revox components, including the B250 amplifier, B226 and B225 CD players, and B215 and B710 tape recorders. I was surprised to find that, although the unit is sold separately, Revox chose not to include the 9-V battery needed to operate it. In any event, after reading the operating instructions that accompanied the remote, I realized that this unit would be great to have if you owned an all-Revox component system, or at least a few other Revox components. For the tuner alone, however, it's a bit expensive.

The reputation that the Swiss have for craftsmanship is certainly confirmed by this tuner. Admittedly, you have to be a pretty serious FM-radio fan to spend the kind of money needed to purchase this elegantly styled and intelligently engineered tuner, but I suspect that there are enough such FM devotees out there to make the B260 the resounding success it deserves to be. *Leonard Feldman*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13

SME V TONEARM AND TALISMAN VIRTUOSO DTi CARTRIDGE

Manufacturer's Specifications

Tonearm

Type: Tapered magnesium tube with fixed headshell and dynamic balance.

Pivot-to-Spindle Distance: 215.35 mm (8½ in.).

Offset Angle: 23°, 38′.

Overhang: 17.8 mm, adjustable ±9 mm by rack-and-pinion shift.

Tracking Error: 0.012°/mm max.; 0° at 66.04 mm and 120.9 mm from record center.

Vertical Tracking Force: 0 to 3 grams, adjustable by 0.1-gram calibrated knob.

Wiring: Headshell, silver litz, user-replaceable; internal, silver litz; external, monocrystal silver with standard five-pin plug.

Effective Mass: 10 to 11 grams, without cartridge.

Weight: 720 grams (1 lb., 9 oz.).

Price: $2,250.

Cartridge

Type: High-output moving-coil.

Stylus: Miniature-shaft van den Hul Type I.

Cantilever: Titanium, diamond-coated.

Output: 1.8 mV into 47 kilohms for 5 cm/S at 1 kHz.

Internal Coil Resistance: 105 ohms.

Recommended Tracking Force: 2 grams.

Frequency Response: 10 Hz to 60 kHz.

Separation: 30 dB at 1 kHz.

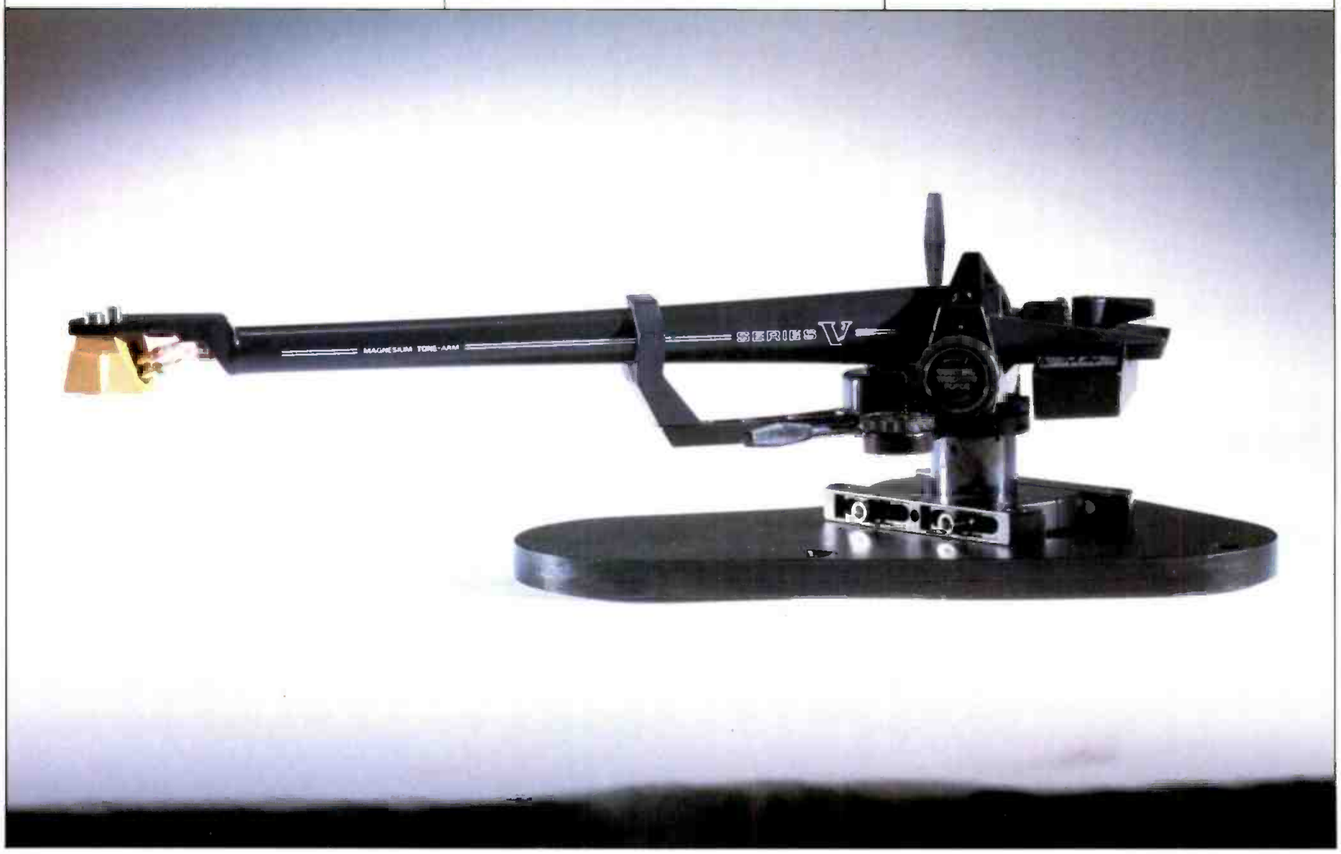
Compliance: 15 × 10^{-6} cm/dyne.

Weight: 7.2 grams.

Price: $1,200; replacement stylus, $720.

Company Address: c/o Sumiko, P.O. Box 5046, Berkeley, Cal. 94705.

(Originally published June 1986)



If you are at all interested in, or even curious about, the trend toward perfection of analog record playback equipment, you will be as interested in reading this report as I was in preparing it. I must admit that I consider the SME V tonearm to be a tour de force in design and manufacture. "Well!" you are saying, "Now I don't need to skip to the end of the report and see what Ed Long really thinks about the SME V tonearm." That's true enough, but if you care to know *why* the SME V works as well as it does and, paradoxically, why you initially might not like the sound which it helps to produce, read on. Besides, the Talisman Virtuoso cartridge is also a part of this report and is interesting enough on its own to be worth reading about.

It was May 1981 when I reported on the SME 3009 III tonearm in conjunction with the Shure V15 Type IV cartridge. A month later, in June 1981, I reported on the Shure MV30HE cartridge wand, which was designed specifically as a plug-in for the SME III. (The SME III was the model previous to the SME V tonearm; there was no SME IV.) If you are interested in comparing the versions III and V to see the change in direction of design exhibited by the SME V, you might want to look at these 1981 reports. A partial comparison between the two is presented here as a "Measured Data" Table.

One thing that has always impressed me, over the years, is the very personal approach exhibited toward the public by SME, a small British company. I think it is due, to a great extent, to SME's Managing Director, Alastair Robertson-Aikman, who is a very personable fellow. He and Reg Eidy, SME's Chief Engineer, have been working on the SME V tonearm for the past five years. At this time, SME is represented in the U.S. by Sumiko of Berkeley, Cal. The Technical Director of Sumiko, David Fletcher, also very personable though something of an iconoclast (Indeed!—*Ed.*), seems to have had some input into the design considerations of the SME V. David was the designer of Sumiko's MDC-800 tonearm and is the person most responsible for the technical aspects of the Talisman moving-coil phono cartridges. Since the latest model of the Talisman line, the Virtuoso (or V, for short), seemed like an excellent match for the SME V tonearm, I used it for the technical and listening tests.

The Talisman V is a new moving-coil design, and while it shares the Direct Field Focus configuration of the Models A, B and S, it is quite different from them in major respects. The Talisman S cartridge was reported on in the September 1983 issue of *Audio*, and you might want to refer to it to see the direction in which the Talisman line of cartridges is heading.

First Impressions

As can be seen from its photograph, the SME V tonearm is very impressive. It reminds me of one of the scale-model starships which have appeared in recent space-epic movies. The fact that the initials SME stand for Scale Model Engineering is purely coincidental, I'm sure. The construction and finish are of the highest quality and make it obvious that one should expect the highest level of performance. The fit of the bearings is excellent, and no play was evident when I tried my push-pull test by gripping the armtube in one hand and the arm pillar in the other. Tapping the armtube produced a very dull sound, indicating that what resonances might be present were well damped and should not color the sound significantly. The SME V tonearm is a dynamic-balance design. This means that after a counterweight is adjusted to balance the tonearm for the cartridge being used, the vertical tracking force and sidethrust compensation force are set by coiled springs, rather than shifting weights, thus keeping the arm in balance even when it is not perfectly level. Two knobs with 0.1-gram calibrations set the tracking force and sidethrust compensation.

There is no finger lift on the headshell. The tonearm is raised or lowered by a viscous-damped mechanism mounted near the main pillar and operated by a control lever.

The Talisman cartridge has one very distinctive feature which I wish were available on all phono cartridges, threaded mounting holes in the cartridge body. That means no

MEASURED DATA

Parameter	Measurements/Comments SME III	SME V
Year of Report	1981	1986
Shape of Armtube	S	Straight
Armtube Material	Titanium	Magnesium
Armtube Damping	Fibrous lining	Constrained-mode
Effective Mass	5.05 grams	10 to 11 grams
Pivot to Stylus	236 mm (9.3 in.)	233 mm (9.2 in.)
Pivot to Rear of Arm	64 mm (2.5 in.)	73 mm (2.9 in.)
Height Adjustment Range	22.2 mm (0.9 in.)	31.8 mm (1.3 in.)
Tracking-Force Adjustment	0 to 2.5 grams	0 to 3.0 grams
Tracking-Force Calibration	Within 0.05 gram	Within 0.15 gram
Cartridge Weight Range	0 to 12 grams	0 to 14 grams
Counterweights	Six	One
Counterweight Mounting	Locked to armtube	Plastic carrier
Sidethrust Correction	String and weight	Spring
Viscous Pivot Damping	Yes	Yes
Damped Arm-Lift Lever	Yes	Yes
Finger-Lift on Headshell	Yes	No
Headshell Offset	24.5°	23.5°
Arm Mounting Slot Required	Standard SME	Standard SME
Overhang Adjustment	Sliding base	Sliding base
Bearing Alignment	Excellent	Excellent
Bearing Friction	Less than 50 mg	Less than 40 mg
Vertical Bearing Type	Knife edge	Ball race
Horizontal Bearing Type	Ball race	Ball race
Lead Torque	Insignificant	Insignificant
External Lead Length	48 in. (1.2 m)	48 in. (1.2 m)
Arm-Lead Capacity	73 pF	85 pF
Arm-Lead Resistance	1.14 ohms	1.20 ohms
Structural Resonances	250 Hz	1.6 kHz
Price	$294	$1,750

Talisman Virtuoso DTi Cartridge

Coil Inductance: 350 µH.
Coil Resistance: Left, 106 ohms; right, 103 ohms.
Output Voltage (47-Kilohm Load): Left, 0.38 mV/cm/S; right, 0.39 mV/cm/S.
Cartridge Mass: 7.7 grams.
Microphony: Very low.
Hum Rejection: Excellent.
High-Frequency Resonance: 25 kHz.
Rise-Time: 13 µS.
Low-Frequency Resonance: 7.5 Hz (in SME V tonearm).
Low-Frequency Q: 2.5.
Recommended Load Resistance: Greater than 5 kilohms.
Recommended Load Capacitance: Less than 1,000 pF.
Recommended Tracking Force: 2.0 grams.
Polarity: Plus, for CD-4 standard.

The SME V tonearm is very impressive. It reminds me of a scale-model starship from a recent space-epic movie.

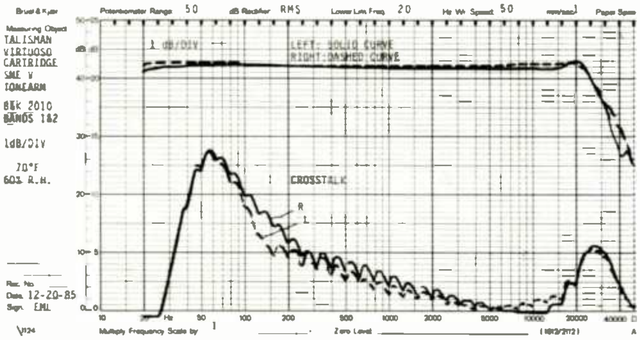


Fig. 1—Frequency response and interchannel crosstalk of the SME arm and Talisman cartridge using B & K 2010 test record.

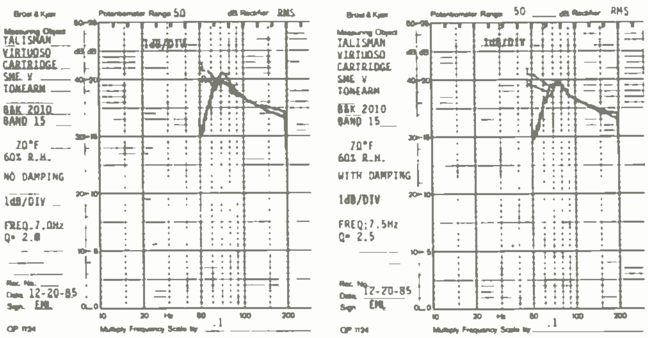


Fig. 2—Low-frequency arm/cartridge resonance is at 7.0 Hz and Q is 2.8 without damping (left); resonance is at 7.5 Hz and Q is 2.5 with damping (right).

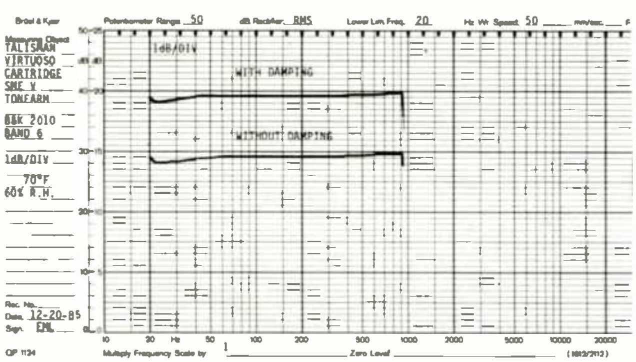


Fig. 3—Slow-sweep check (20 to 900 Hz) for tonearm resonances, with damping (upper curve) and without (lower curve).

separate nuts or washers must be held in place while the screws are lined up, started, and tightened, all while holding the cartridge in place. I would like to publicly thank St. Anthony for the many small hardware pieces which he has found for me, over the years, after they had seemingly fallen into some black hole during the intricate cartridge-mounting process.

Features

The armtube of the SME V tonearm is pressure die-cast of magnesium, which has a very good stiffness-to-weight ratio and good inherent damping. The hollow arm tube tapers in both thickness and cross-section along its length, and it extends from the headshell to the counterweight, passing through the pivot bearings without any joints or breaks. The internal damping of the armtube is accomplished by using a technique called constrained-mode damping (more on that later), which has proven very effective in reducing the type of mechanical-energy buildup most prevalent in thin-walled structures.

There are two holes with steel inserts in the headshell for mounting the phono cartridge. Since there are no slots in the headshell, the necessary stylus overhang must be adjusted by moving the whole tonearm in relationship to the turntable's center spindle. This stylus overhang is required for all pivoted tonearms to correct for tracking error across the record surface. The lack of slots does not allow the angle of the cartridge to be adjusted to correct for any misalignment of the stylus on the cantilever. With a "Line Contact" type of stylus, this can be a problem, because it means that the left and right groove walls will not be traced at precisely the same instant. This effect shows up in the phase-versus-frequency plot which I have been showing for a number of years. I am not certain how the interchannel time offset affects the total sound quality, but in the limited experimenting I have done, it seems to have an effect upon the precision of the sound image in the upper frequency range. In any case, the quest to eliminate any odd effects from the reproduced sound should make it a matter of some concern.

The counterweight system has a die-cast casing with weights inside. It is shaped so that it can be placed very close to the pivots, allowing the tonearm to maintain a very low effective mass. Cartridges weighing between 4 and 14 grams can be balanced by rotating a thumbwheel, which adjusts the position of the counterweight. After the cartridge is balanced, the counterweight can be locked in place by turning a lever. The tracking force can be set to as much as 3 grams by turning the calibrated knob.

The vertical tracking angle can be adjusted in one direction while playing a record, by turning the VTA screw, which raises the tonearm pivots. To lower the pivots, the main pillar must be pushed down again. Adjustment of the VTA is also helped by the two white lines which run the length of the armtube and by a special template. The 10-mm diameter horizontal and 17-mm diameter vertical bearings are captive in precision races, and located in the same plane as the record surface, to negate the effects of warp wow. The tonearm can be moved with respect to the center spindle by a rack-and-pinion system which uses a special horizontal

I wish all cartridges had the Talisman's threaded mounting holes, which eliminate the need for tiny nuts and washers.

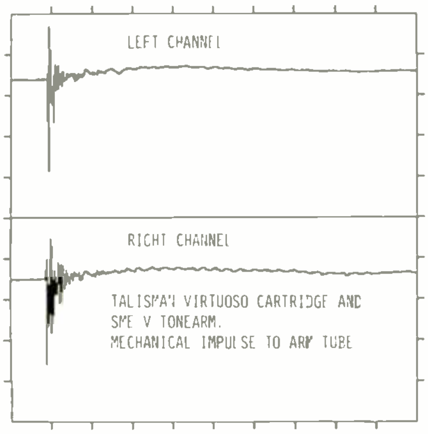


Fig. 4—Output of arm/cartridge when mechanical impulse was applied to armtube. Total period of measurement was 20.47 mS.

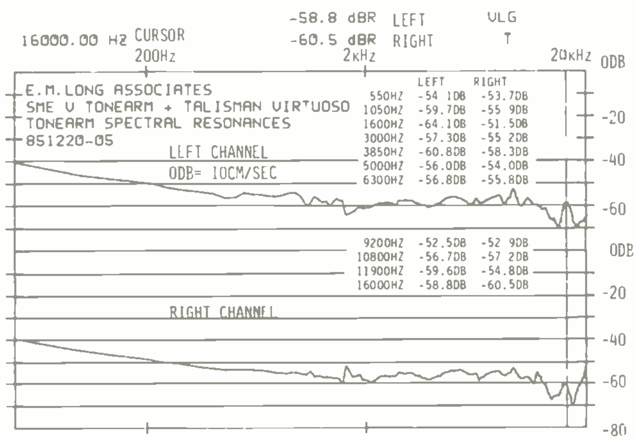


Fig. 5—Output (averaged) of arm/cartridge due to 16 mechanical impulses applied to armtube. This is an excellent result, indicating good energy damping by the tonearm.

Fig. 6—Interchannel phase difference of arm/cartridge using pink noise from B & K 2011, band 7.

tracking angle key. The HTA key allows the tonearm to be adjusted so that the stylus will overhang the center spindle by the correct amount. SME has used this technique before but never with such an elaborate and precise mechanism.

The tonearm damping system, which consists of a trough of viscous fluid and a paddle, is also a refinement of the previous system used on the SME III tonearm. The amount of damping in the horizontal plane is increased or decreased by lowering or raising the paddle with an adjustment screw. The four color-coded cartridge attachment leads can be changed; the ones supplied were by van den Hul. The internal armtube wiring and the external phono cables are also specially made for SME by van den Hul. The bottom of the arm pillar is fitted with a viscous-damped, right-angle plug which can be rotated almost a full 360°. This allows the phono cables to exit the turntable base wherever it is most convenient and solves the problem of how to anchor them properly.

As a last note about this tonearm's features, I must compliment SME for the quality and comprehensiveness of the instruction manual. It is a gem.

The body of the Talisman V cartridge is machined from a solid piece of aluminum. As mentioned previously, the mounting holes are drilled and tapped right into the body. A high-coercivity neodymium-iron magnet is used in the Talisman's Direct Field Focus system to achieve a very high output voltage. In fact, the output of the Talisman V is about eight times that of the Talisman S, so it needs no step-up device. This means not only lower system cost but that the signal degradation inevitable with a step-up is eliminated. The Talisman V can be operated directly into a 47-kilohm, magnetic phono-cartridge input. In fact, any load impedance above 5 kilohms is acceptable, and input capacitance also has negligible effect. The cantilever of the Talisman Virtuoso which I tested is titanium, with a diamond overlay (as denoted by its DTi designation), which gives it added stiffness. (A Talisman Virtuoso B, with boron cantilever, is also available, for $850.) The stylus is a van den Hul Type I design, precision-cut from a small cross-section diamond shaft. A damper having very low hysteresis is used in order to maintain good time and amplitude response for better spatial-information recovery.

Measurements and Listening Tests

Figure 1 shows the amplitude versus frequency response and also the crosstalk versus frequency for the SME/Talisman combination; the results are very good. The response at 20 kHz is particularly interesting since it verifies comments by some members of the listening panel that the sound was very open and clear in the upper range. The high-frequency resonance at 25 kHz, due to the interaction of the compliance of the record groove with the effective mass of the stylus tip, is verified by the rise in the crosstalk. The rise in the crosstalk at 60 Hz is an artifact of the B & K 2010 test record; the actual crosstalk for the SME/Talisman combination is much lower.

The low-frequency resonance, caused by the interaction of the tonearm's effective mass and the cartridge's compliance, is shown in Fig. 2, with and without the low-frequency damping system activated. The Q is very low for either

The panel had to learn to appreciate this pair's tightly controlled bass, which was lower but more realistic than with the reference system.

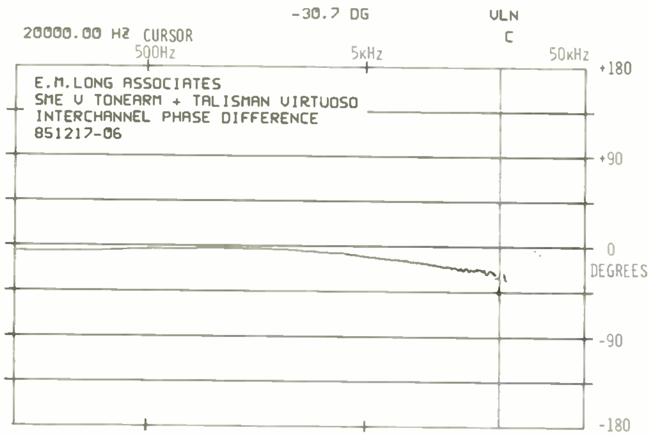
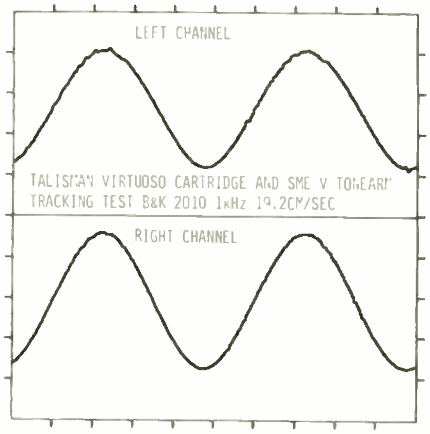


Fig. 7—Interchannel phase difference of arm/cartridge as a function of frequency (B & K 2011, band 7, pink noise). Phase difference at 20 kHz (cursor position) is 30.7°, equivalent to only 4.3 µS of interchannel delay.

Fig. 8—Tracking of arm/cartridge with a 1-kHz test tone from the B & K 2010 at 19.2 cm/S.



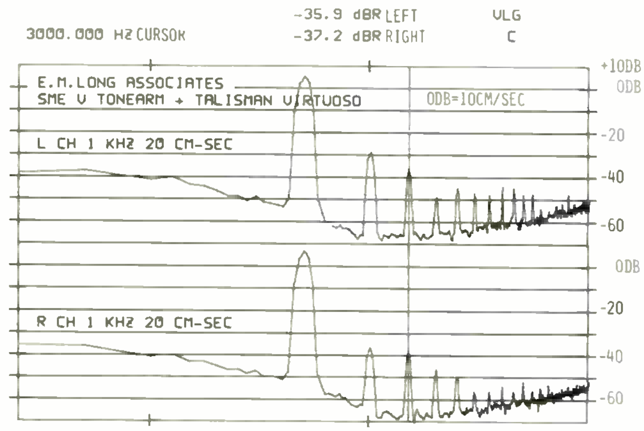


Fig. 9—Spectral analysis of the cartridge output when reproducing the test signal of Fig. 8. The third harmonic (at the cursor position) is 1.6% in the left channel, 1.4% in the right.

condition, but it must be remembered that I have taken much care in centering the test record, which reduces the side-to-side swaying of the tonearm. The viscous damping system is most effective for this horizontal motion, so it would be even more useful under normal conditions, with the record less carefully centered, than would appear from Fig. 2. The panel's comments about the very tightly controlled character and extension of the bass register were a bit difficult to sort out. Due to better damping of its low-frequency resonance, the SME/Talisman combo had less bass output than the reference system's, though perhaps a bit more extended. This confused some panel members a bit. The reference system was preferred by some panel members when reproducing drums and double bass because, as they commented, "the bass sounds fuller." This is another paradox of the SME/Talisman performance, since careful listening to the sound of bass drum on orchestral recordings conv nced me that it was more realistic.

The smooth curves of Fig. 3 indicate the control of mechanical resonances in the SME V tonearm. The viscous damping system does not have any visible effect upon the response, but there aren't any major uncontrolled vibrations which would show its effect anyway. This lack of substantial resonances, which might color the sound, can be verified easily by tapping the armtube with a pen or a small screwdriver. I am convinced that it was this very lack of coloration which caused some members of the listening panel to comment that the SME/Talisman combination was more reticent in its quality than the reference system. This is what I meant earlier, when I said that you might not like the sound which the SME V tonearm helps to produce. The lack of coloration in the SME can cause other tonearms to be preferred, at least in short-term listening. I have seen this before, especially in the case of a tonearm which caused a very euphonic coloration due to its very loose pivots!

The results of applying a series of mechanical impulses to the armtube are presented in Figs. 4 and 5, which show the amplitude versus time and the frequency spectrum of the energy of the SME V/Talisman Virtuoso combination. These results indicate the excellent damping of energy by the SME's tapered magnesium armtube and the internal damping system. As I mentioned before, the technique used is called constrained-mode damping. This method places the actual damping material in direct contact with the surface from which the energy is to be removed. The damping material is then covered with, and its movement constrained by, a thin layer of stiff material. The energy is dissipated in the form of heat as it tries to move between the surface to be damped and the constraining layer.

Figure 6 shows the interchannel phase difference, displayed on the screen of a Nicolet Explorer III digital storage oscilloscope. Figure 7 shows the phase versus frequency

Even if you can't afford the SME and Talisman right now, you should audition them to hear how good records can sound.

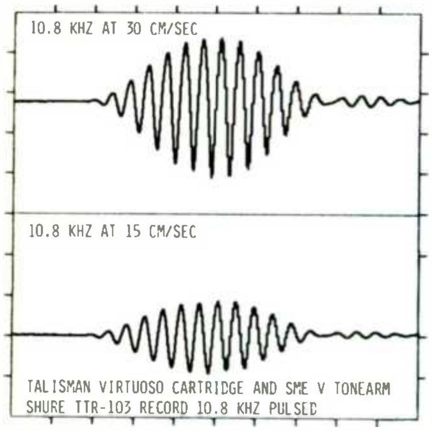


Fig. 10—Output with 30- and 15-cm/S, 10.8-kHz pulse test, Shure TTR-103 test record.

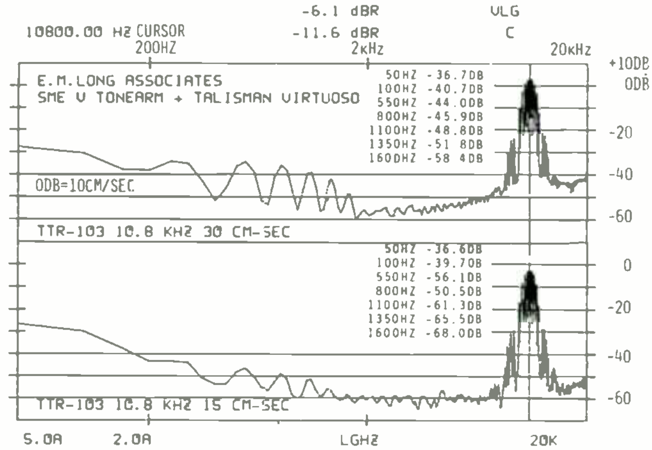


Fig. 11—Spectral analysis of distortion products from signals shown in Fig. 10. Distortion at 50 Hz is 1.5% for 30-cm/S modulation.

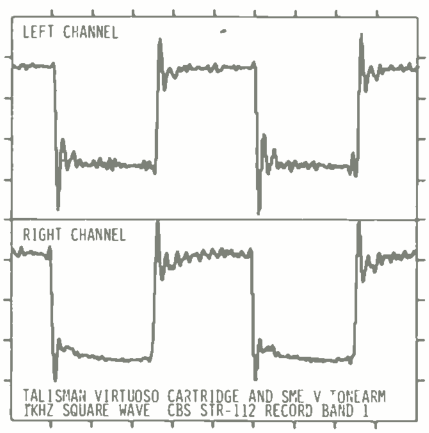


Fig. 12—Cartridge output from 1-kHz square wave, CBS STR-112 test record.

for 64 samples of the same pink noise signal used for Fig. 6. This is a phase transfer function, obtained using a Nicolet 660A-2D Fast Fourier Transform analyzer, and represents the difference in phase between the left and right channels. The very slight phase difference between the channels indicates that the ability to present a very stable stereo image is excellent. The comments of the listening panel confirmed this by indicating that both the SME/Talisman and the reference system presented superb stereo images.

The SME/Talisman combination has excellent tracking ability, as shown by Figs. 8 and 9. Figure 8 shows amplitude versus time of the left- and right-channel outputs while reproducing the second-highest-level 1-kHz band on the B & K 2010 test record. The waveform has only a tiny amount of jitter. The frequency spectrum of this waveform is shown in Fig. 9, and the distortion is very low for this high a level. The eighth and ninth harmonics show an increase; this may account for some of the brightness which was heard when reproducing a trumpet.

The ability of the SME/Talisman combination to reproduce high frequencies with a seeming effortlessness is verified by the information shown in Figs. 10 and 11. The output of the left channel, while reproducing a 10.8-kHz tone burst at 30 and 15 cm/S, is shown in Fig. 10; its symmetry and lack of compression, even at the higher level, is very good. The aspect of performance most often cited as an area of superiority over the reference system was the way that the SME/Talisman combination reproduced the midrange and high frequencies. The words "transparency" and "clarity" were often used by members of the listening panel to describe the sound.

The responses to 1-kHz square waves, Fig. 12, are excellent and indicate that, while time delay of the high frequencies with respect to low frequencies is present, it is very slight. The relationship of fundamentals and overtones will therefore be correct over a wide range of frequencies.

Conclusions

Both the SME V tonearm and the Talisman Virtuoso cartridge incorporate some very useful and innovative features. Most notable about the SME are its tapered, one-piece magnesium armtube and integral headshell; special van den Hul internal wiring, and the constrained-mode internal damping in the armtube. The Talisman's most remarkable features are, I feel, its solid aluminum body, with integral drilled and tapped mounting holes, and the diamond-coated titanium cantilever. Together, the SME V tonearm and the Talisman Virtuoso cartridge make an excellent combination; anyone who is looking for the finest reproduction of analog discs should try to audition it. Even if you can't afford to buy these components right now, you should listen to them to hear just how good records can sound. *Edward M. Long*

14

EMINENT TECHNOLOGY TONEARM 2 AND TALISMAN VIRTUOSO B CARTRIDGE

Manufacturer's Specifications

Tonearm

Type: Straight-line, air-bearing arm with fixed headshell and interchangeable, tapered armtube.

Pivot-to-Stylus Distance: 7⅜ in. (18.7 cm).

Tracking Error: 0°.

Effective Mass: Vertical, 7 grams; horizontal, 25 to 35 grams.

Height Adjustment: ¾ in. (19 mm).

Vertical-Tracking-Angle Adjustment: Arcuate, ±2° from center of range.

Overhang Adjustment: ½ in. (12.7 mm).

Cueing: Adjustable-height, eccentric-bar mechanism.

Azimuth Adjustment: ±2° from vertical.

Tracking-Force Range: 0 to 7 grams.

Pivot Damping: Vertical, none; horizontal, motion damped at counterweight assembly.

Bearing Surface Area: 6.8 square in.

Overall Arm Weight: 14.25 oz. (405 grams).

Lead Wire: Van den Hul linear-crystal silver; oxygen-free litz wire optional.

Capacitance: 40 pF.

Resistance: 0.9 ohm.

Materials: Hard-anodized aluminum armtube and bearing spindle; high-modulus carbon-fiber composite armtube joint; glass-fiber and mineral-filled composite bearing housing.

Air Filter: Disposable, 2 to 3 years average life expectancy.

Price: $850 including air pump.

Company Address: 508 Cactus St., Tallahassee, Fla. 32301.

Cartridge

Type: High-output moving coil.

Stylus: Van den Hul Type I.

Cantilever: Boron rod.

Output: 1.8 mV for 5 cm/S at 1 kHz.

Recommended Load Impedance: 47 kilohms.

Internal Impedance: 105 ohms.

Frequency Response: 15 Hz to 32 kHz, ±2 dB.

Channel Separation: 35 dB at 1 kHz.

Channel Balance: 0.5 dB.

Compliance: 15×10^{-6} cm/dyne.

Recommended Tracking Force: 2 grams, ±0.25 gram.

Weight: 7.5 grams.

Price: $856; replacement stylus, $471.

Company Address: c/o Sumiko, P.O. Box 5046, Berkeley, Cal. 94705.

(Originally published February 1987)

The quest for a way to retrace the grooves of a phonograph record as perfectly as possible always leads back to "linear" or "straight-line" tracking. The original master lacquer, which is used to produce the record, is made by a cutterhead which is driven across its surface in a straight line, so it seems obvious that the best way to retrace a record would be by using a straight-line method. This is easier said than done, and the fact that most tonearms in the world are pivoted certainly bears this out. It is much easier to produce a low-friction conventional arm, which has bearings only at its pivot, than to produce a good straight-line arm which must maintain low friction along a bearing that runs across the record from the outside to the inside grooves.

Many really great innovations are the result of someone's desire to produce something that seemed impractical, at least to the "bean counters" or money-oriented mentalities of the time. The Eminent Technology Tonearm 2 is the result of Bruce Thigpen's desire to produce a linear-tracking tonearm which would overcome the biggest problem of such tonearms: Friction. It was a long time coming and makes an interesting story.

When Bruce was still a student at Florida State University, in the late 1960s, he met Lew Ekhart. Lew was developing air bearings which could be used by students to perform physics experiments. When Bruce mentioned that such a bearing would probably be just the thing to make a really great turntable, Lew told him that he had made one and asked if he would like to see it. I can just imagine Bruce's excitement at the prospect!

In the early 1970s, both Lew and Bruce were employed by the Wayne Coloney Company, which worked on military contracts. The company was looking for other products to manufacture, and Lew's air-bearing turntable was developed. (This turntable is now being produced by the Mapleknoll Company.) In 1982, Bruce decided to start his own company and use his expertise in air-bearing technology to produce a linear-tracking tonearm. The air bearing, which is the main feature of the design, results in almost zero friction.

The Eminent Technology Tonearm 2 on a SOTA Star Sapphire turntable.

MEASURED DATA

Eminent Technology Tonearm 2

Pivot-to-Stylus Distance: 7.0 in. (17.8 cm).
Pivot-to-Rear-of-Arm Distance: 2.75 in. (7.0 cm).
Overall Height Adjustment: 0.8 in. (2.0 cm).
Tracking-Force Adjustment: 0 to 5 grams.
Tracking-Force Calibration: None; separate gauge required.
Cartridge Weight Range: 0 to 12 grams.
Counterweights: Lead; two 4.8 grams, two 14.9 grams.
Counterweight Mounting: Leaf spring for horizontal, solid vertical.
Sidethrust Correction: None needed.
Lifting Device: Lever, no damping.
Headshell Offset: Not needed.
Overhang Adjustment: Slot in armtube (see text).
Bearing Alignment: Excellent.
Bearing Friction: Very low.
Lead Torque: Slight near inner and outer grooves.
Arm-Lead Capacitance: 47 pF.
Arm-Lead Resistance: 1.9 ohms.
Structural Resonances: 1,100, 1,550, 2,650, and 5,700 Hz.
Base Mounting: Single center hole.

Talisman Virtuoso B Cartridge

Coil Inductance: Left, 310 μH; right, 280 μH.
Coil Resistance: Left, 103.8 ohms; right, 102.3 ohms.
Output Voltage: Left, 0.36 mV/cm; right, 0.38 mV/cm.
Tracking Force: 2.0 grams used.
Cartridge Mass: 7.5 grams.
Microphony: Excellent.
Hum Rejection: Excellent.
High-Frequency Resonance: Left, 20.8 kHz; right, 22.7 kHz.
Rise-Time: 15 μS.
Low-Frequency Resonance: 10 Hz (in Eminent Technology Tonearm 2).
Low-Frequency Q: Less than 1 (in ET2 tonearm).
Response to Load Capacitance: Unaffected by normal input capacity.

The first version of the tonearm lacked some of the refinements of the present version. Many of these were the result of input from Edison Price, the owner of an engineering company in New York City, who bought one of Thigpen's first tonearms. Price applied some of his mechanical engineering expertise to add some features which he thought would make the tonearm easier to adjust and use. Bruce welcomed these design improvements and has incorporated them into the present version of the tonearm. Most small companies rely on good word-of-mouth advertising by satisfied customers; Edison Price has proved to be the ultimate satisfied customer, because he actively promotes the tonearm and has even set up a demonstration listening room in his own facility in New York.

With its air pump on, the arm feels slippery, its friction reduced to practically nothing.

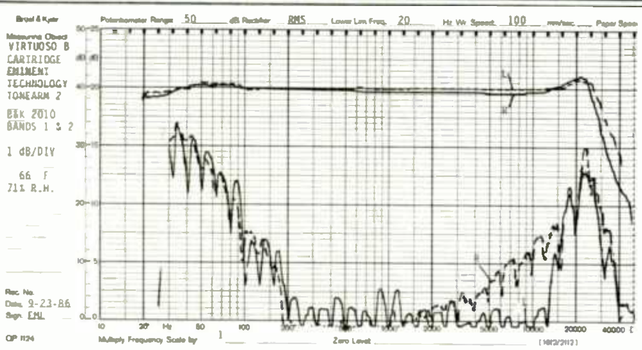


Fig. 1—Frequency response and crosstalk, Talisman Virtuoso B cartridge in Eminent Technology Tonearm 2, using B & K 2010 test record.

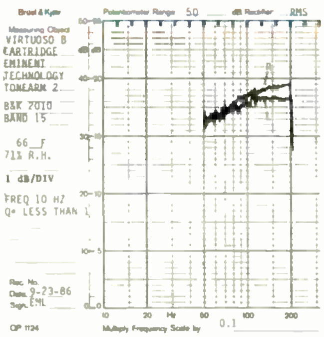


Fig. 2—Low-frequency tonearm/cartridge resonance is at 10 Hz with a Q of less than 1, B & K 2010.

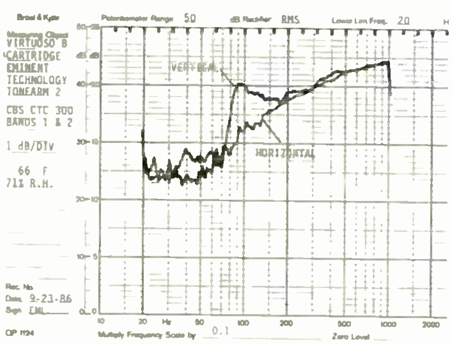


Fig. 3—Response to vertical and horizontal modulation from 2 to 100 Hz (slow sweep). Note differences between horizontal and vertical response due to arm and counterweight construction (see text).

First Impressions

"Solid" and "massive" were the first adjectives that came to my mind when I first encountered the Eminent Technology Tonearm 2 (which I will refer to as the ET2, for the sake of brevity). It is finished entirely in black, with white markings. The horizontal bearing is composed of a long spindle or tube which slides back and forth inside a rather large, solid housing. This housing is called the bearing manifold because it is the device into which air is pumped. When the air pump is turned off, there is a bit of friction between the horizontal bearing tube and the bearing manifold as the tube is moved back and forth. I was surprised that there is so little play in the bearing when the air pump is off, and even less when the pump is on. But the main surprise, to me, was how slippery the bearing felt when the air pump was on. The friction was reduced to practically nothing.

Unlike some other linear-tracking tonearms, whose horizontal bearings are located out over the record, the ET2 has its main bearing set back along the right edge of the turntable. Because of this, it doesn't have to be moved out of the way to change a record, but it does require that the armtube that carries the cartridge be longer and therefore have greater effective mass. The armtube of the ET2 still has a lower effective mass in the vertical plane than most pivoted tonearms.

I was at first both impressed and disheartened by the large number of adjustments offered by the ET2. Being able to vary many parameters allows the tonearm to be adjusted perfectly, but it also means that great care must be taken. As I proceeded with my evaluation of the ET2, I began to appreciate what could be accomplished by carefully manipulating each of the interrelated adjustments. Eminent Technology offers some optional set-up fixtures that can make the job easier, but if you have any trepidation, the dealer should be able to set up the ET2 on your turntable.

Features

The main feature of the ET2 is the air bearing, which has 6.8 square inches of surface. The moving part of the air bearing, which is called the spindle tube, is made of 6061-T6 hard-anodized aluminum. It has a 0.6110-inch outside diameter and a 0.014-inch wall thickness, and is machined to a tolerance of ± 0.0003 inch. The stationary part of the air bearing, which is called the manifold assembly, is made of carbon-fiber composite material. The armtube is 5052-T5 hard-anodized aluminum with an outside diameter of 0.5 inch and a wall thickness of 0.035 inch. The cartridge end is tapered to a flattened surface and has a compressed Teflon insert. The armtube also has a vinyl sheath on the outside and closed-cell foam inside to control resonances. The interconnect plug for the phono leads and the fastening screw for the armtube are located near the vertical pivot; this lowers their contribution to the effective vertical mass. The armtube is interchangeable and performs a function similar to that of the headshell on other tonearms. The armtube slides over a solid-aluminum joint insert, located at the end of the bearing spindle, which expands to meet the inside of the hollow armtube when the single fastening bolt is tightened. Because the joint insert is relatively heavy, it too is located near the vertical pivot to reduce its contribu-

The large number of adjustments on the arm is impressive and, at first, disheartening. This allows perfect alignment but requires great care.

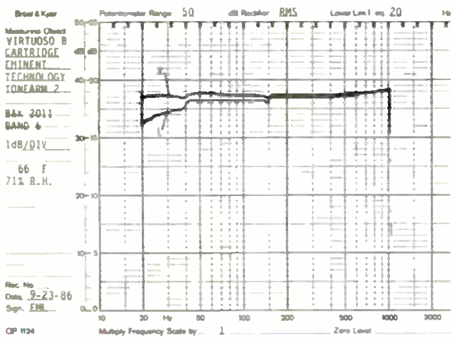


Fig. 4—Slow sweep from 20 Hz to 1 kHz. Note resonances at about 40 and 150 Hz.

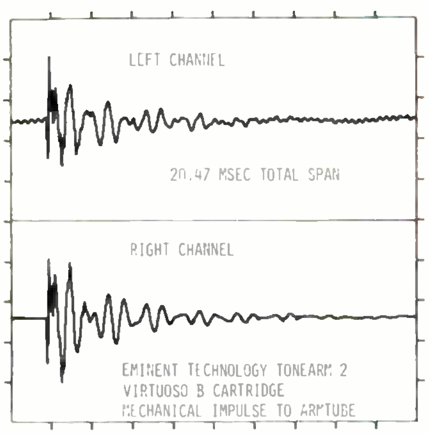


Fig. 5—Output vs. time of arm/cartridge when mechanical impulse was applied to armtube, with arm on rest.

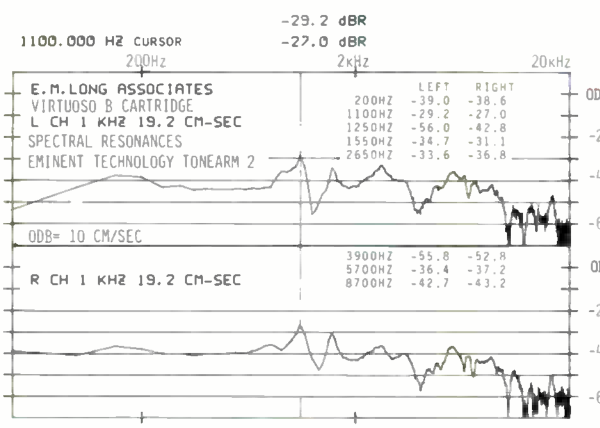


Fig. 6—Spectral output (average) of arm/cartridge due to 16 mechanical impulses applied to armtube.

tion to the effective vertical mass. To allow for different cartridge heights, the spindle bearing end of the joint insert can be located in any of three vertical positions, and it is locked in place by a bolt. A slot in the armtube, over the joint insert, allows the overhang to be adjusted over a 0.5-inch range to allow for different cartridges' stylus-to-mounting-hole distances. Because this slot is slightly wider than the locking screw, the armtube can be rotated slightly. This allows adjustment of the vertical azimuth (the stylus' perpendicularity to the record, as viewed from the front).

The bearing manifold is indirectly attached to the mounting post, which is also made of carbon-fiber composite material, through a vertical gear system which allows the vertical tracking angle to be adjusted. Since a straight up or down movement of the bearing manifold would cause the position of the stylus to move back or forth, the tracking-angle control lever moves the bearing manifold through an arc. This causes the overhang adjustment to remain constant. The VTA can be adjusted while a record is playing if the turntable suspension permits it. A supplied snap-on gauge allows adjustments as fine as 0.1°; the company offers an even higher precision gauge at extra cost.

The counterweight, located at the end of the spindle bearing away from the armtube, is an I-beam with graduated markings. These are for reference only and are not calibrated in grams. The beam is attached to the spindle bearing by a leaf spring, which is compliant only in the horizontal plane. To balance the arm, two 15-gram and two 5-gram lead weights are supplied, allowing adjustment from 5 to 40 grams in 5-gram steps. For fine dynamic balancing, the counterweight assembly can be raised or lowered and moved in and out along the I-beam track. After balancing the arm, tracking force is set by sliding the counterweight assembly along this track.

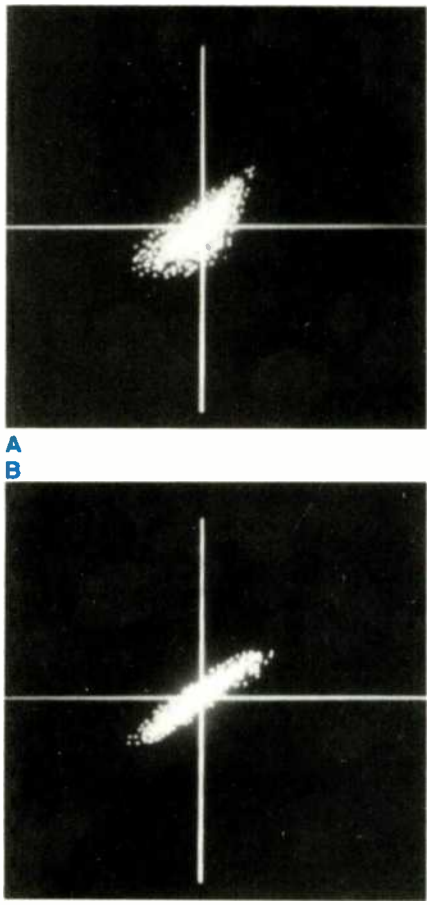
The lead wires inside the armtube are van den Hul linear-crystal silver, with oxygen-free litz wires available on special order. The cartridge ends of these leads are attached to gold-plated connectors which can be slipped over the cartridge pins. The wires are color-coded to match the coding on most cartridges. The other end of these wires exit the armtube near a slot in the joint insert and are attached to a five-pin plug. After the armtube is slipped over this joint insert, the plug can be inserted into the socket in the end of the bearing spindle. The leads then continue across the inside of the bearing spindle and exit on the side near the counterweight assembly. They are then looped down through a hole, which must be drilled in the tonearm mounting board, and routed to the outside of the turntable base. There they are terminated to two gold-plated phono jacks which are mounted on a small plastic box. This box can be attached to the rear of the turntable base.

The air pump is a bit noisy. However, since it is supplied with a 24-foot hose, it can be located in another room. That is what I did during the listening tests.

Measurements and Listening Tests

The Talisman Virtuoso B cartridge was used with the Eminent Technology Tonearm 2 to obtain the data shown in the technical measurements (Figs. 1 through 14) and also during the listening tests. I did use some other cartridges

This arm and cartridge put out less bass than other combinations do, but after listening a while, the others sound bass-heavy!



A
B

Fig. 7—Interchannel phase difference, using pink noise from B & K 2011, band 7. With arm aligned so stylus is on the radius that runs at right angles to the armtube, slight phase errors appear (A); with arm adjusted so stylus is 0.125 inch ahead of this radius, channels are better synchronized (B).

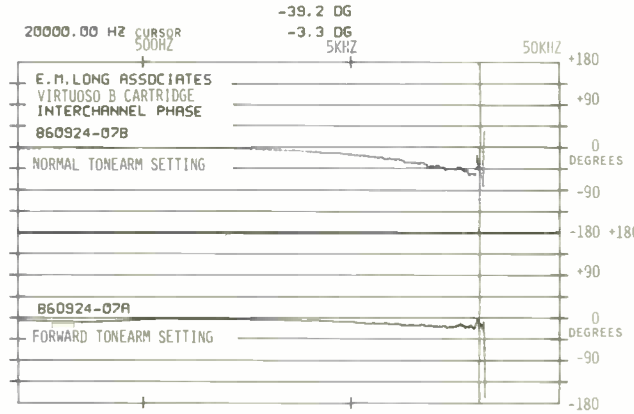


Fig. 8—Interchannel phase difference vs. frequency for B & K 2011, band 7, pink noise. Top curve, for normal stylus location on radius at 90° to armtube, shows phase error of 39.2° (equivalent to 5.4 μS) at 20 kHz; bottom curve, for stylus 0.125 inch ahead of this radius, shows error reduced to 3.3° (0.5 μS).

during the technical measurements to double-check some results. The Virtuoso B is a high-output moving-coil cartridge with a boron rod cantilever, which accounts for the "B" designation. It was operated directly into the phono preamp with a 47-kilohm resistive load shunted by about 200 pF of capacitance from the leads and phono preamp input. The Virtuoso B has a moderate compliance, and it worked very well with the ET2 tonearm.

Figure 1 shows the frequency response and separation of this arm/cartridge combination. Most of the drop in separation at the low frequencies is due to crosstalk on the B & K 2010 test record. At higher frequencies, there is a bit more crosstalk from the left channel into the right than vice versa, but it is still very good, less than 30 dB at frequencies up to a little above 6 kHz. The image stability was considered by the listening panel to be excellent.

Figure 2 confirms that the low-end response of this arm/cartridge combination does indeed drop off below 20 Hz, as suggested by Fig. 1. The slow sweep from 5 to 20 Hz provided by band 15 of the B & K 2010 test record shows that a very well-damped resonance occurs somewhere around 10 Hz with a Q of less than 1. The consensus of the listening panel was that the ET2 had slightly less bass than my reference system but that the bass was of exceptionally good quality. For some panel members, the difference in timbre between the tuba and trombone was more easily distinguishable on the reference system, and the double bass sounded deeper and more powerful.

Figure 3 is a test of the ET2/Virtuoso B combination's response to vertical and horizontal (lateral) modulation from 2 to 100 Hz. The response of the combination in the horizontal plane is very low, with a tiny, low-Q resonance at about 4 Hz and another at about 9 Hz. The response to vertical modulation is more pronounced, with a resonance of about 9.5 Hz and a Q of about 1.9. This confirms that the ET2 does have more mass in the horizontal than in the vertical plane and that the counterweight, which has more compliance horizontally than vertically, is working properly.

Output from a linear arm such as this can remain very symmetrical under difficult conditions. The cartridge, not the arm, sets the tracking limits.

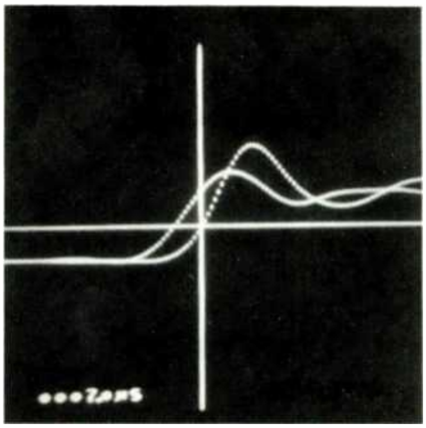


Fig. 9—Time delay between right and left channels. Right channel leads left by 7 μS.

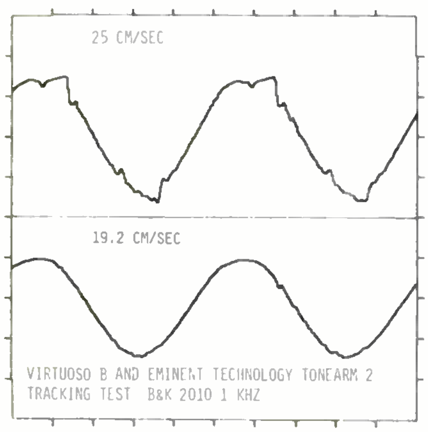


Fig. 10—Tracking of arm/cartridge with 1-kHz test tones at 25 cm/S (highest level on B & K 2010, top) and 19.2 cm/S (bottom).

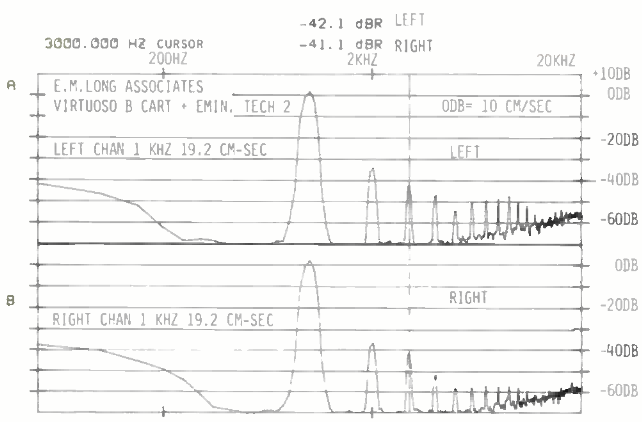


Fig. 11—Spectral analysis of the cartridge output when reproducing the 19.2-cm S signal of Fig. 10. The third harmonic (at the cursor position) is 0.78% in the left channel and 0.88% in the right.

Figure 4 shows resonances in the ET2 at about 40 and 150 Hz. This indicates that there are mechanical interruptions to the flow of energy which cause cancellations to occur at these frequencies. I do not know exactly what effect the air-bearing interface has upon this flow of energy, but because it acts as a barrier, it doesn't allow energy to be transferred and dissipated in the tonearm board, as is the case with other tonearm designs. These resonances tend to correlate with most panel members' perception of a slightly brighter sound from the ET2/Virtuoso B combination for voice, trumpet, and strings. This is not to be taken as a negative statement, because most panel members rated the combination as being slightly preferable to the reference system when reproducing these sounds.

Another test for the effect of delayed mechanical energy is the mechanical impulse test, shown in Figs. 5 and 6. Figure 5 shows the output versus time for a mechanical impulse applied to the tonearm near the cartridge. There are some oscillations with a time period of about 2 mS which exhibit alternating polarity reversals. This is a clue that reflected energy is present. Figure 6 shows the spectral output due to a series of mechanical impulses applied to the ET2 near the cartridge. Averaging the output produced by 16 separate impulses imparts a greater reliability to the test and therefore greater validity to the results. The graph shows the reinforcement and cancellation of energy and the relative levels at the peaks. One panel member commented that rapid piano tones were a little more distinct when reproduced by the reference system as compared to the ET2, and this would correlate well with the data of Figs. 5 and 6.

Figures 7A and 7B show an interesting phenomenon for which I have not discovered a satisfactory explanation. Figure 7A shows the left- versus right-channel output for the pink noise of band 7 of the B & K 2011 test record. If the channels were perfectly synchronized, with no time delay between them, the pattern would be a straight line at 45°. Moving the cartridge forward in the armtube, creating an overhang of 0.125 inch between the stylus and the record radius that runs at right angles to the armtube, produced the output shown in Fig. 7B. This more closely approximates the 45° line that should result when the channels are synchronized. Figure 8 shows the difference in interchannel phase versus frequency for the two cartridge positions, averaging 16 samples of the B & K 2011 test record. Further tests with other cartridges and overhang settings indicate that it is not simply a cartridge or overhang problem. Tests with other records (which had music and test signals) made with different cutting systems also indicate that the type of cutterhead isn't the answer either. The dimensional discrepancies which cause these interchannel phase or synchronization errors are extremely tiny, however. The 7-μS error at 1 kHz shown in Fig. 9 (the leading edge of the 1-kHz square wave on band 1 of the CBS STR-112 test record) represents a difference between the left and right channels of 0.00014 inch for a record rotating at 33⅓ rpm with the stylus in a groove at a 5.75-inch radius. The dimensional difference for the same signal at an inner groove of the record is even smaller, because the information density is greater. All of this may seem trivial, but since interchannel differences are

The Eminent Technology arm feels very different and takes time to get used to, but its performance is well worth it in the end.

being shown for Compact Disc players, I thought that a comparison might be interesting. I think further investigation by others might lead to some interesting insights and might even allow further advances in the quality of analog record reproduction.

Figure 10 shows the output of the combination when trying to track the high-level 1-kHz test tones of the B & K 2010 test record. There is some mistracking at the highest level (25 cm/S), but it is very symmetrical. I think this is interesting because it shows how a linear-tracking tonearm, which neither has nor needs any sidethrust correction, can produce a very symmetrical output under difficult conditions. The limit to tracking ability, in this case, is the cartridge, not the tonearm. The Virtuoso B cartridge is very good, however; the lower trace, representing the output at 19.2 cm/S, and the spectrum of the distortion (shown in Fig. 11) verify this.

The output of the ET2/Virtuoso B combination produced by the 10.8-kHz tone burst of the Shure TTR-103 test record is shown in Fig. 12. Even at the highest level, which is for a 30-cm/S groove modulation, the results are quite good. The spectra produced by both the 30- and 15-cm/S levels are shown in Fig. 13. The distortion levels listed in the figure are very low and indicate that the high-frequency tracking capability of the combination is excellent. No adverse comments were made by any of the listening panel members regarding the sound of cymbals, bells, or other difficult-to-reproduce, high-pitched instruments. In fact, one panel member said that the sound of these instruments from both the ET2/Virtuoso B combination and the reference system were really fabulous.

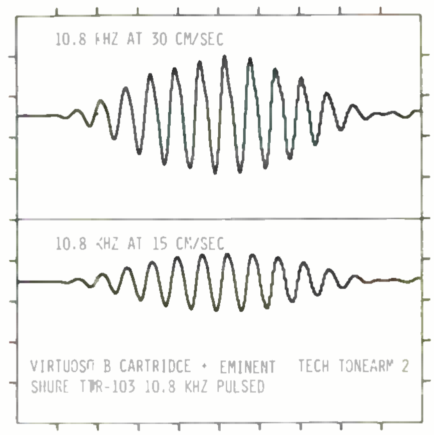
Figure 14 shows the output of the combination for the square wave on band 1 of the CBS STR-112 test record. The characteristic decrease in output immediately following the sharp initial transient is normal for a cartridge which exhibits the type of roll-off shown previously in Fig. 1. This type of characteristic is also associated with the phase or time delay exhibited by any band-limited device or circuit. It would be interesting to determine whether any difference could be heard if a phase-correction circuit were inserted between the cartridge and the loudspeakers.

Conclusions

My initial trepidations about the apparent complexity of the Eminent Technology Tonearm 2 were completely gone by the time I finished the technical measurements and the listening tests. The Talisman Virtuoso B cartridge is practically identical to the Virtuoso/DTi, which I reported on in the June 1986 issue of *Audio*, but less expensive. The ET2 does have a feel that's different from the usual pivoted tonearm, and it takes a little time to get used to it. However, it is certainly worth the effort, because the ET2's performance is excellent. The level of low bass from the ET2/Virtuoso B combination is less than that from other tonearm/cartridge combinations, but after listening to this setup for a while, the others sound bass-heavy! The final judgment of the listening panel and myself is that the Eminent Technology Tonearm 2 and the Talisman Virtuoso B cartridge make a fantastically accurate and musically pleasant combination. If you love music, you will love them.

Edward M. Long

Fig. 12—Output from 30- and 15-cm/S, 10.8-kHz pulse test, Shure TTR-103 test record.



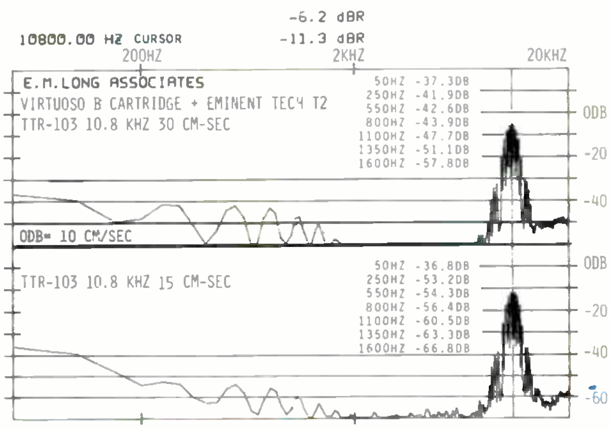
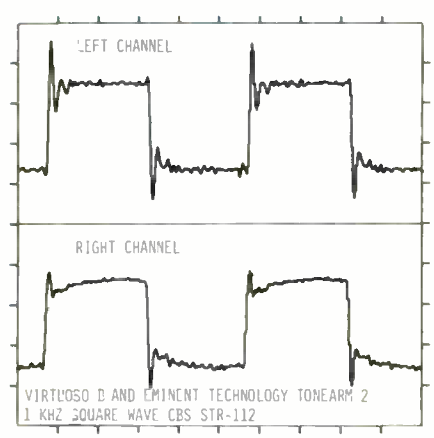


Fig. 13—Spectral analysis of distortion products from signals shown in Fig. 12 (average of 16 samples at each level). Distortion is highest (1.5%) at 50 Hz on 15-cm/S level.

Fig. 14—Output from 1-kHz square wave, using CBS STR-112 test record.



15

WELL TEMPERED TURNTABLE

Manufacturer's Specifications
Drive System: Belt.
Motor Type: 24-pole synchronous.
Speeds: 33⅓ and 45 rpm.
Dimensions: With optional dust cover, 19 in. W × 15½ in. D × 7 in. H (48.26 cm × 39.37 cm × 17.78 cm).
Weight: Approximately 38 lbs. (17.3 kg).
Price: Without arm, $1,075; with Well Tempered arm, $1,795; dust cover, $100.
Company Address: c/o Transparent Audio Marketing, Rte. 202, Box 117, Hollis, Maine 04042.
(Originally published July 1988)

After Bill Firebaugh had designed the Well Tempered tonearm, he decided to apply some of his ideas on viscous damping to the design of a turntable. The absorption of energy from external vibrations is something that most turntable designers are concerned about; however, most of them concentrate their efforts on reducing vibration problems in the low-frequency range. They do this by using a suspension system employing strategically located springs. The compliance of these springs is chosen so as to resonate with the mass of the turntable at some very low frequency, usually below 10 Hz. A suspension of this type tends to reduce the effects of outside vibrations and shocks on the reproduced sound.

One way of reducing vibration effects is to use very compliant springs to suspend a relatively low mass. Other designers use less compliant springs but make the suspended mass very large, which tunes the system to the same range. In this case, however, the Q is usually higher, so some method must be used to damp the springs.

Continued on page 120

WELL TEMPERED TONEARM AND VAN DEN HUL MC-ONE CARTRIDGE

Manufacturer's Specifications

Tonearm

Type: Pivoted, with adjustable cartridge mount and viscous damping.

Pivot-to-Stylus Distance: 9 in. (22.9 cm).

Overall Length: 11⅜ in. (28.9 cm).

Effective Mass: 10 grams.

Arm Tube: Stainless steel, sand-filled.

Price: $825.

Cartridge

Type: Medium-output moving coil.

Stylus: Van den Hul Type I.

Cantilever: Boron rod.

Output: 0.45 mV for 5 cm/S at 1 kHz.

Tracking Force: 1.5 grams recommended; 1.3 grams minimum.

Estimated Vertical Compliance: 20×10^{-6} cm/dyne.

Mass: 7.3 grams.

Recommended Arm Mass: 6 to 12 grams.

Load Impedance: 15 ohms minimum.

Frequency Response: 20 Hz to 20 kHz, ±0.75 dB.

Channel Separation: 40 dB at 1 and 10 kHz; approximately 20 dB at 20 kHz.

Price: $1,075; stylus retipping, $537.50.

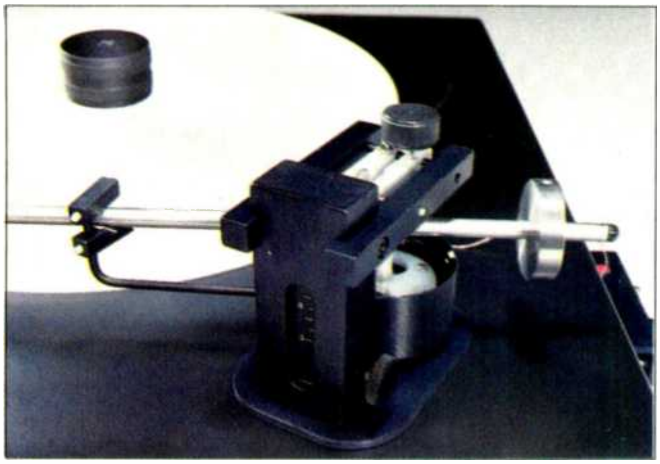
Company Address: c/o Transparent Audio Marketing, Rte. 202, Box 117, Hollis, Maine 04042.

(Originally published July 1988)

One of the pleasant things about doing technical reports for *Audio* magazine is being able to investigate, in detail, the products of some very innovative minds. Most of the improvements in sound reproduction from phonograph records is due to certain individuals' dissatisfaction with the state of the art. Each of them looked at the way

Photographs: ©1988, Bill Kouirinis

The arm produced a subdued "deh" or "dah" when tapped, a sign that it would not introduce much coloration.



things were being done and said. "There must be a better way." Their improvements either found their way into others' products or spurred their colleagues to even greater innovation. We have all benefited from the work of people like Lou Souther, Ivor Tiefenbrun, Alastair Robertson-Aikman, Bernard Jacobs, Dave Fletcher, John Michell, Bruce Thigpen, Steve McCormack, Herbert Papier, Joe Grado, A. J. van den Hul, and others. (If I have left anyone out—and I most assuredly have—I can always blame the Editor!)

The Well Tempered tonearm is the result of the innovative thinking of Bill Firebaugh, and even its appearance is radical enough to win him the "Iconoclast of the Year" award. He started his quest for the perfect record player by taking apart his early AR tonearm, designed by Ed Villchur (there's a name I missed). As Firebaugh told me, "I had burned my bridge. I had no tonearm now. There was no turning back!" He liked the AR's viscous-damped vertical bearing, but he found himself adjusting it too often and decided that there must be a way to make it more consistent. As a mechanical engineer for a large aerospace company in Southern California, he had the background to tackle a job like that, but, as often happens during a quest for perfection, the trail he took resulted in something very far removed from the original AR design.

When Bill came to my lab from Los Angeles to set up his turntable and tonearm (with the van den Hul MC-One moving-coil cartridge) I asked him if he had ever seen the Gray Professional tonearm, which was made in the 1950s for broadcast studios. He said he had only heard about it. I had owned one, and I described to him the problem of keeping the viscous-damped bearing adjusted. The Gray used a single half ball at the bottom of the tonearm post; the half ball was seated in a matching cup which contained viscous fluid. The glitch was that the arm used to settle, push the viscous fluid out of the way, and allow the ball to come in direct contact with the cup, thus negating the fluid's damping effect. The Gray had to be adjusted by pulling the tonearm up and holding it while the viscous fluid slowly settled back down toward the bottom of the cup. Thus, you needed a good deal of patience to complement your dedication to quality sound! The Well Tempered tonearm eliminates this tedious adjustment problem by suspending the bearing so that it never settles down into the viscous fluid. (If I had thought of that years and tears ago, I might still have my old 16-inch Gray!) But there are other features of the Well Tempered that make tonearms of yesteryear the quaint curiosities they are.

The MC-One is a moving-coil cartridge made by A. J. van den Hul, and it incorporates his patented stylus design, which is shaped to trace the difficult high frequencies pro-

MEASURED DATA

Well Tempered Tonearm

Pivot-to-Stylus Distance: 9.375 in. (238 mm).
Pivot-to-Rear-of-Arm Distance: 3.0 in. (76 mm).
Overall Height Adjustment: 1.25 in. (32 mm).
Tracking-Force Adjustment: 0 to 6 grams.
Tracking-Force Calibration: None; separate gauge required.
Cartridge Weight Range: 3 to 12 grams.
Counterweights: 20.9-gram aluminum and 61.5-gram steel.
Counterweight Mounting: Direct to armtube, with nylon set screw.
Sidethrust Correction: Caliper dial varies string-pivot spacing.
Pivot Damping: Viscous fluid in large cup.
Lifting Device: Aluminum finger lift attached to cartridge.
Headshell Offset: Cartridge offset adjustable in mount; no headshell.
Overhang Adjustment: Slots in cartridge mount.
Bearing Type: Armtube suspended by two strings.
Bearing Alignment: Adjustable pivot point.
Bearing Friction: Viscous fluid in cup.
Lead Torque: Very low.
Arm-Lead Capacitance: 25 pF, left and right.
Arm-Lead Resistance: 1.1 ohms, left and right.
External Lead Length: None supplied.
Structural Resonances: 550, 1200, 3800, and 4900 Hz.
Base Mounting: Single hole.

van den Hul MC-One Cartridge

Coil Inductance: 120 µH, left and right.
Coil Resistance: 15.7 ohms, left and right.
Output Voltage: Left, 0.10 mV/cm/S; right, 0.11 mV/cm/S.
Tracking Force: 2.0 grams recommended.
Recommended Load Resistance: 40 ohms or more.
Response to Load Capacitance: Unaffected by normal input capacitance.
Cartridge Mass: 7.25 grams.
Microphony: Very low.
Hum Rejection: Excellent.
Rise-Time: 11 µS.
High-Frequency Resonance: 33.3 kHz.
Low-Frequency Resonance: 10 Hz (in Well Tempered tonearm).
Low-Frequency Q: 1.67 (in Well Tempered tonearm).
Polarity: Plus, for CD-4 standard.

The vertical bearing design ensures equal up and down motions when the tonearm is tracking vertical warps.

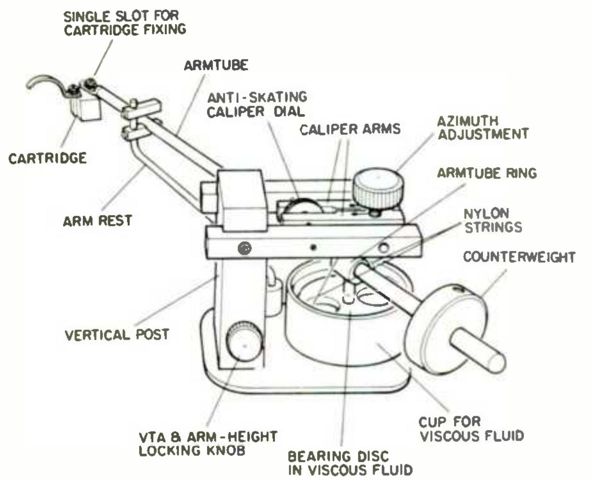


Fig. 1—The Well Tempered tonearm showing the locations of various design features

duced by the chisel-shaped styli used in cutting records. While this stylus is available for license by other cartridge manufacturers, most of them opt for a simpler, generic stylus with a long-ellipse contact area.

First Impressions

By its very appearance, the Well Tempered tonearm is different from any other arm. The main bearing is suspended by two nylon strings and sits in a bath of viscous fluid. Aluminum, stainless steel, and various types of engineering plastics are used in the arm's fabrication. The main armtube is one continuous piece, from the cartridge fixing point to the end which holds the sliding counterweight.

I usually check for bearing integrity by holding the main arm post in one hand while trying to pull, push, and twist the armtube. The design of the Well Tempered arm precludes this, since the bearing is free to move in the viscous fluid. I did tap the armtube, and the sound varied from "deh" (as in delicious) when tapped at the cartridge end, to a very subdued "dah" (as in "ah! That's nice!") when tapped near the pivot. This test told me that the amount of sound coloration introduced by the armtube would indeed be very low.

The finish is black and natural stainless steel and is very good quality. The lack of a headshell is also unusual; the cartridge must be mounted by a single screw to an aluminum extension fitted into the end of the armtube. There are no conventional calibrations anywhere on this tonearm, with the exception of a scale on the main arm post which can be used to set the VTA of the cartridge. The caliper-like device used to set the sidethrust or anti-skating force is something I have never seen before on a tonearm.

Features

As I describe various features, you can refer to Fig. 1, a line drawing of the Well Tempered tonearm. The first thing to notice is that this is basically a unipivot design. Most unipivot bearing designs hold the armtube from below, but this arm is unique in that the pivot, while below the armtube, is suspended from above by strings. The exact position of the main bearing is somewhere near the center of the nylon bearing disc, which is suspended by two nylon strings; it is not fixed in position, as is true with ordinary unipivot designs. The exact position of the horizontal bearing is affected by the azimuth adjustment, while the vertical position will be affected by the height of the cartridge used and the VTA setting. With most cartridges, the vertical bearing center can be positioned slightly above the record surface. This causes the stylus to move up and down equally when tracking vertical warps, which is desirable.

The nylon disc, which has two large holes in it, sits in a bath of viscous fluid that just covers its surface. This viscous fluid provides excellent damping, especially at the usual low-frequency resonance caused by the mass of the tonearm and the compliance of the phono stylus. A short aluminum post extends upward from the nylon disc and connects to a thin but wide aluminum ring with a hole in it. The armtube is securely fastened in this ring. The armtube, which is 0.256 inch in diameter and 0.010 inch thick, is made from 0.250-diameter stainless steel tubing and is 11½ inches long. It is filled with fine-grain sand to damp out any resonances and weighs about 23 grams, including the cartridge fitting and end cap.

The phono cartridge mounting is by a single screw, since the fitting at the end of the armtube is about ¼ inch wide and has only a single slot. This is a trade-off: The rigidity of the cartridge mounting has been compromised to keep the effective mass of the tonearm low. The aluminum finger lift is attached directly to the cartridge, through the cartridge's remaining mounting hole. Of course, this can be eliminated, if desired, to keep the effective mass as low as possible, but since there is no other easy way to raise and lower the tonearm, I left it attached.

The offset adjustment for the phono cartridge is made by rotating the cartridge on its single mounting screw. The offset angle and the overhang are set by using a plastic template which slips over the turntable spindle. The overhang is adjusted by rotating the whole tonearm base around the main pillar, which fits up inside the large, rectangular vertical post.

Many interdependent adjustments must be made to get all this right. As mentioned before, the vertical azimuth, which is adjusted by turning the knob directly over the armtube, affects the position of the horizontal bearing; it also affects the overhang. The sidethrust correction is adjusted by turning the caliper dial, which can be seen sticking up between the vertical post and the azimuth knob. Turning this dial varies the distance between the two aluminum caliper arms to which the nylon strings are attached, thus adjusting the sidethrust correction. Even at the minimum setting, with the beams close together, there is some sidethrust force because of the way the strings are attached. I found this setting worked best with the van den Hul cartridge.

Loosening the knob at the bottom of the vertical post allows you to set the proper tonearm height for the cartridge chosen; it also is used to adjust the VTA. The adjustment must be made by sliding the tonearm up and down by hand.

The van den Hul cartridge and Well Tempered arm had a distinctive sound that correlated well with the results of my measurements.

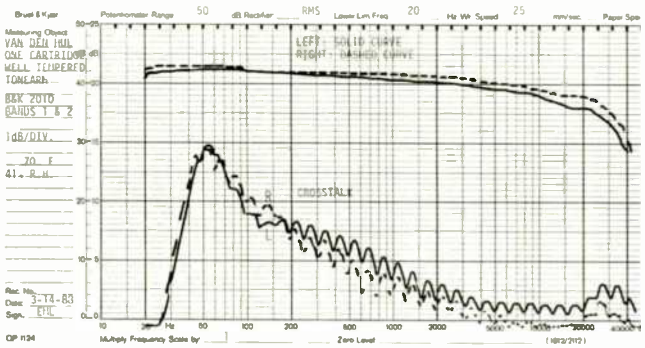


Fig. 2—Frequency response and crosstalk of the van den Hul MC-One phono cartridge in the Well Tempered arm, using B & K 2010 test record.

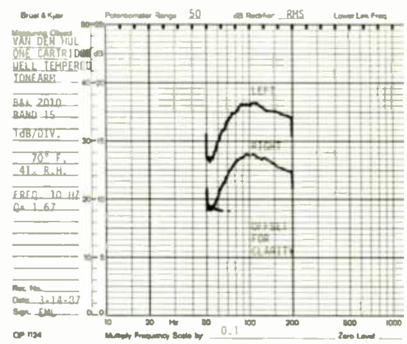


Fig. 3—Low-frequency tonearm cartridge resonance is at 10 Hz. Its Q is 1.67, which is very good. (Curves offset for clarity.)

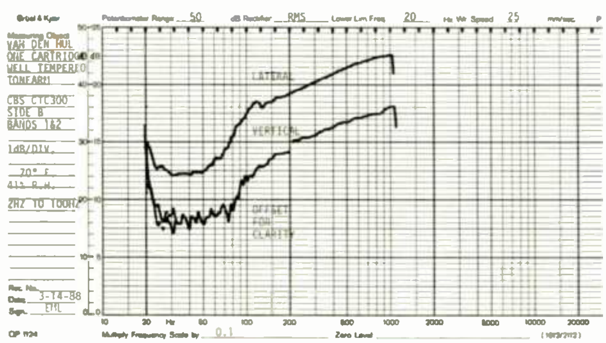


Fig. 4—Response to vertical and horizontal modulation from 2 to 100 Hz (slow sweep). Note that arm cartridge resonance is almost completely damped. (Curves offset for clarity.)

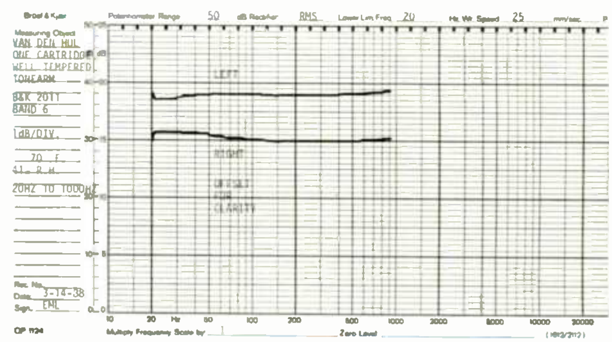


Fig. 5—Slow sweep from 20 Hz to 1 kHz, to check for structural resonances in the arm. Resonances can be seen but are very subdued. (Curves offset for clarity.)

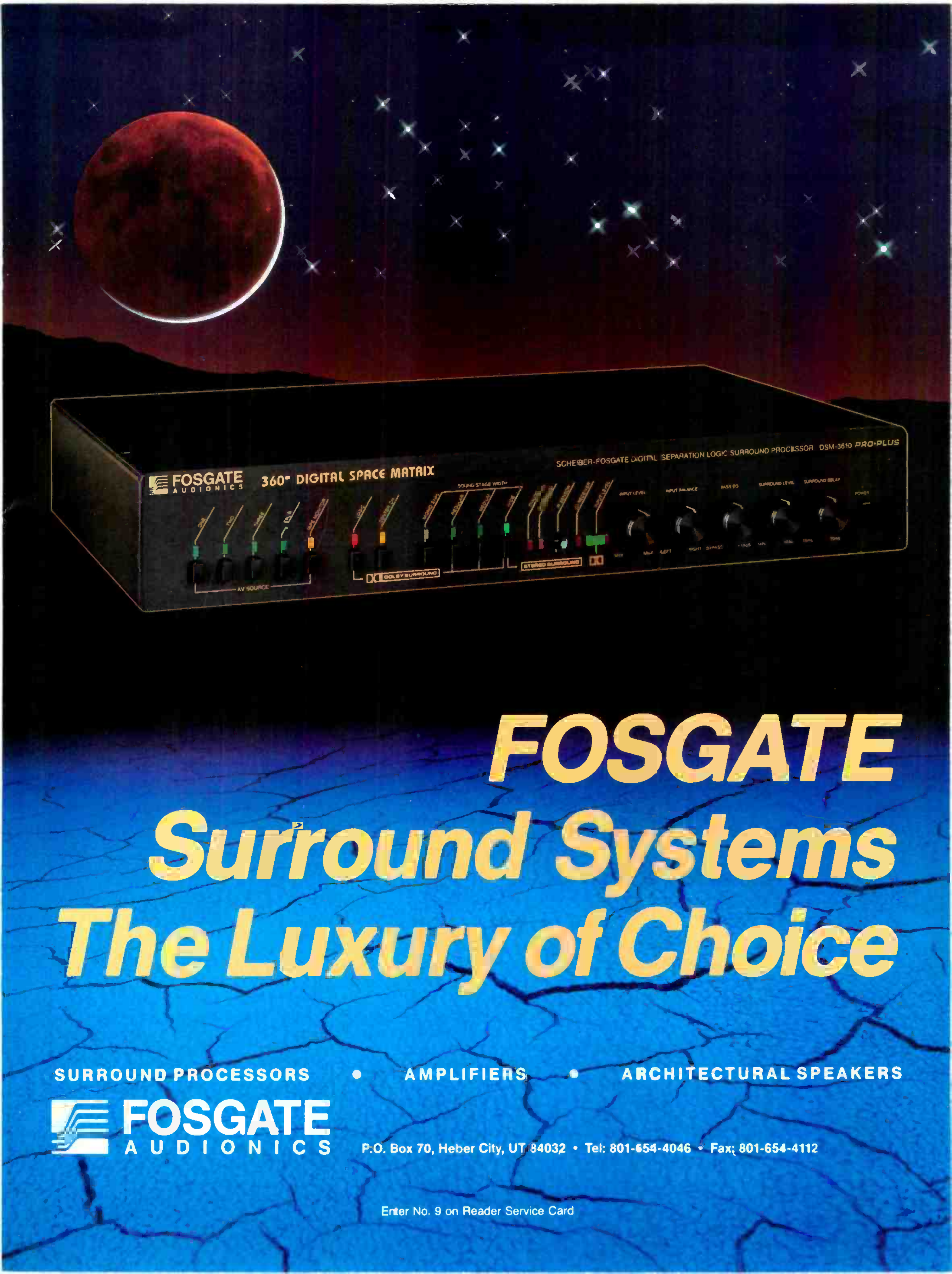
A calibrated scale on the arm post (not shown in Fig. 1) can be seen through an opening on the side of the rectangular vertical post. The scale is marked from +3 to −3, with zero in the center. Once at the proper height, a line can be drawn on the post in line with the zero. (The arm's height may cause interference with some turntables' dust covers.)

Vertical tracking force is adjusted by sliding one of the two supplied weights along the rear of the armtube. The counterweight should be as close to the pivot as possible, so choose the heaviest practical weight. The circular counterweight should be locked by the nylon set screw. A separate tracking-force gauge must be used, since there are no calibrations on the armtube or counterweights.

Gold-plated connectors are used at the cartridge end of the fine Litz-wire phono leads, which exit the armtube just in front of the armtube ring (visible under the rear horizontal bar). These leads are twisted together and have plenty of slack, so they can be dressed for lowest lead torque. Two gold-plated phono sockets and a five-way binding post for grounding are mounted to an extruded aluminum channel, which can be attached on the rear of the turntable base.

Measurements and Listening Tests

The correlation between the technical measurements and the written comments made by members of the listening panel seems to be very good. The combined "sonic signature" of the Well Tempered tonearm and van den Hul cartridge was different enough from that of the reference system to make this easy. Remember, the reference system is not intended to represent the ultimate goal to match; only live sound could serve that purpose. The sound of the system being evaluated can be judged better or worse than the reference, but that is not its purpose. The reference merely acts as a point from which the listening panel members can rate the systems from 0 to −5 and make comments about the sound as they perceive it for each of the 12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The low crosstalk figure helps account for the good imaging of this arm/cartridge pairing.

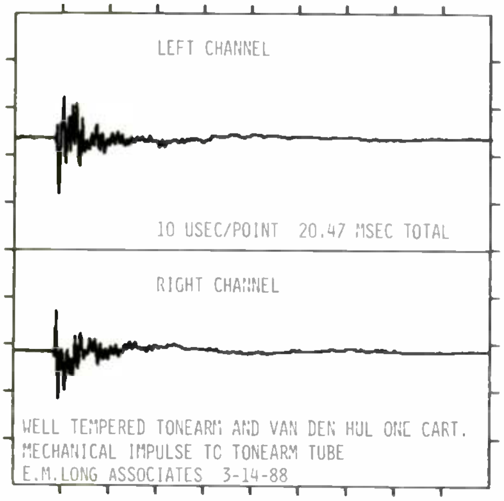


Fig. 6—Output vs. time of arm cartridge when mechanical impulse was applied to armtube, with arm floating.

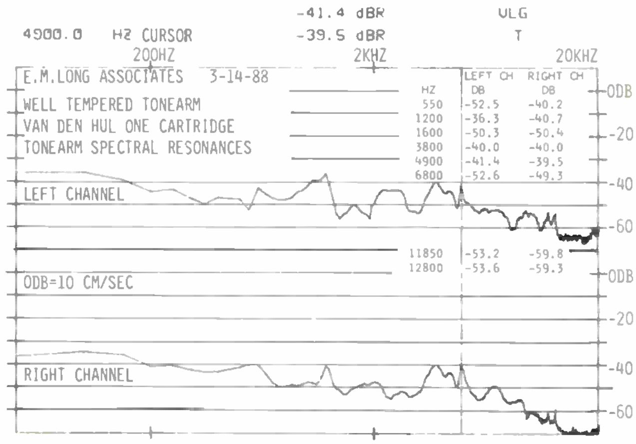


Fig. 7—Spectral output (averaged) of arm cartridge due to 16 mechanical impulses applied to armtube. Most of the energy is in the middle register, which could add to perceived brightness.

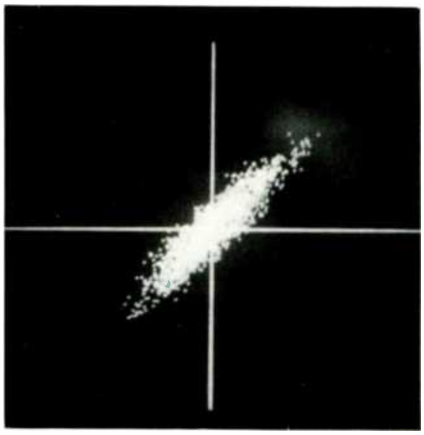


Fig. 8—Interchannel phase, using pink noise from B & K 2011, band 7.

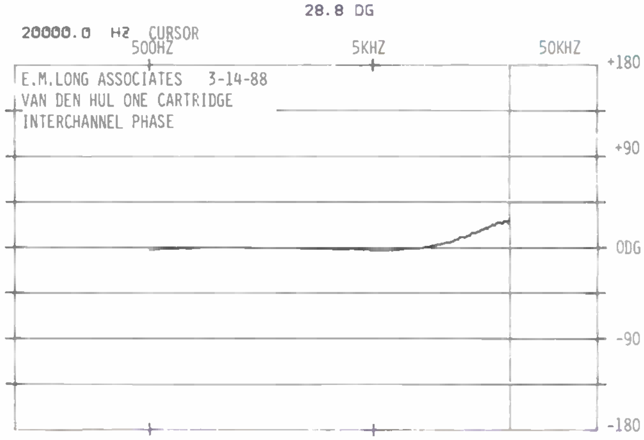


Fig. 9—Interchannel phase difference vs. frequency for B & K 2011, band 7, pink noise. Phase difference at 20 kHz is 28.8° (4.0 μS).

musical selections played. The reference system should be a very good one—and it is—but its main value comes from the fact that it is a known, measured, and repeatable quantity. The rating and comments can be looked at in light of these measured differences.

All panel members commented that the reference system sounded sharper on the sounds of cymbals, brass and string overtones, etc. Figure 2 shows the amplitude versus frequency response of the WT/vdH combination. There is an apparent roll-off of the higher frequencies, which would account for these comments. These remarks were not negative in tone; some wrote "smoother highs on cymbals," "sonorous," etc. The crosstalk measurement usually indicates very distinctly the high-frequency resonance of the cartridge, but it is barely visible for the van den Hul, being somewhere in the region around 30 kHz. This means that the resonance is well controlled, which is good. I made other tests for this and determined that the high-frequency resonance is at 33.3 kHz. The amount of crosstalk is very low, which helps account for the good comments made about the imaging of this arm/cartridge combination. Comments about the balance of the sound being "forward" might be explained by the shape of the curves, which show more output in the fundamental range of instruments and voice, but other data correlates with this as well.

Figure 3 is the left- and right-channel response from 5 to 20 Hz, with the curves offset for clarity. It shows the low-frequency resonance caused by the effective mass of the

The ability of this arm and cartridge to track high-level mid-frequencies proved exceptional.

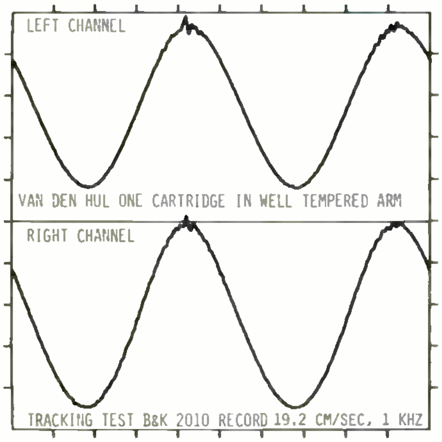


Fig. 10—Tracking of arm cartridge with 1-kHz test tones at 19.2 cm S (B & K 2010). a level most cartridges find difficult to track. A small jitter is visible at the top of the waveform.

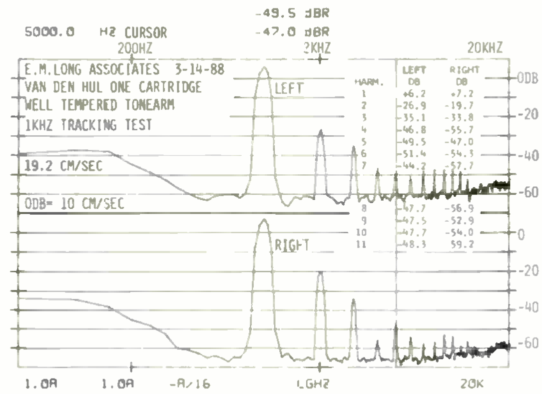


Fig. 11—Spectral analysis of the cartridge output when reproducing the 19.2-cm S signal of Fig. 10. The fifth harmonic (at the cursor position) is 0.19% in the left channel and 0.20% in the right.

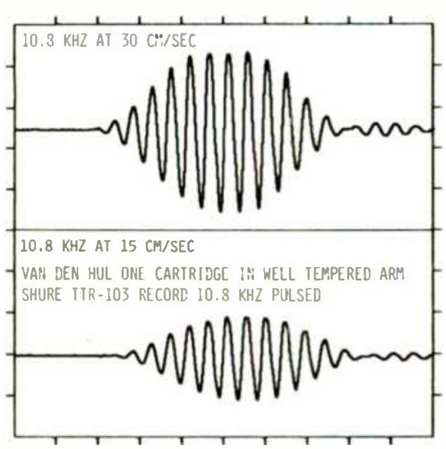


Fig. 12—Output from 30- and 15-cm S, 10.8-kHz pulse test, using Shure TTR-103 test record.

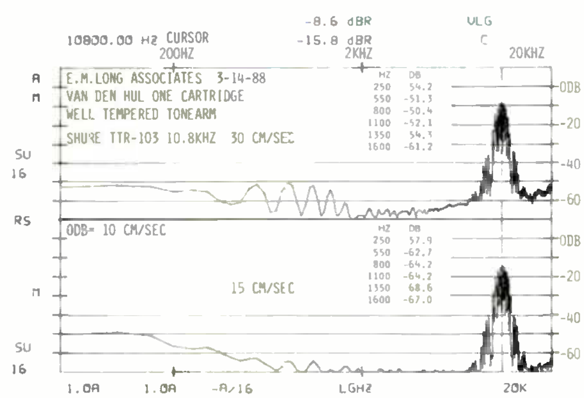


Fig. 13—Spectral analysis of distortion from signals shown in Fig. 12 (average of 16 samples at each level). The level at 250 Hz is 0.2%, which is very good. Output at 30 cm S is +8 dB above the 10-cm S, 0-dB reference level.

Well Tempered tonearm interacting with the compliance of the stylus of the van den Hul cartridge. The resonance is at 10 Hz and is very "well tempered" or damped by the viscous fluid at the pivot of the tonearm. Other tonearms with damping near their pivots have not shown results as good as this. The quality of the bass is excellent with this arm and cartridge. One panel member said it provided "tighter" sound, while the overall comments and ratings indicate a tie with the reference system. Figure 4 shows the combination's response from 2 to 100 Hz for lateral and vertical groove modulation. The resonance is more apparent in the lateral mode, which indicates that the damping is greater in the vertical plane of tonearm movement.

Figure 5 shows the response of the Well Tempered tonearm/van den Hul cartridge combination to a slow sweep from 20 to 1,000 Hz. Any resonant "rattles" caused by loose fittings will show up during this test. There are very tiny indications of resonances, especially in the right channel, but nothing really severe.

Figure 6 shows the WT/vdH's response to a mechanical

Some listeners liked this pairing very much. It was very precise, dry, and analytical, and tied the reference for clarity.

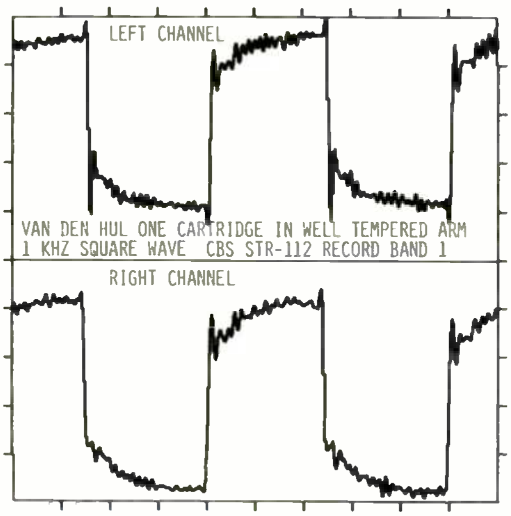


Fig. 14—Output from 1-kHz square wave, using CBS STR-112 test record.

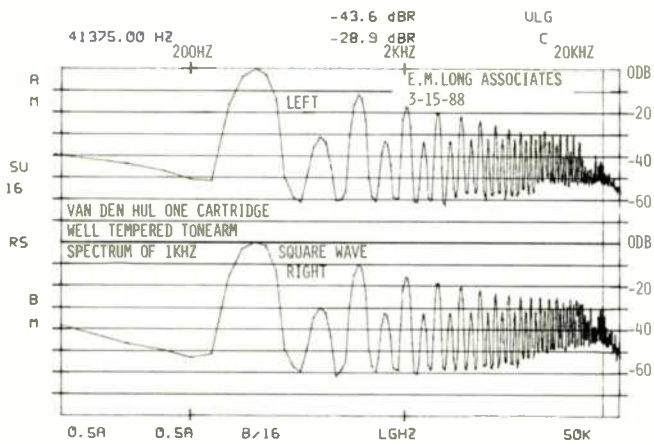


Fig. 15—Spectral analysis of 1-kHz square wave (STR-112).

impulse. The internal damping of the tonearm is good: The output decays rapidly without showing any serious delayed reflections. Figure 7 shows the spectrum of the response to an average of mechanical impulses applied to the tonearm Some panel members commented that the sound of individual violin and orchestral strings was "brighter"; this might be due to the fact that the Well Tempered tonearm shows more energy in the range between 3.5 and 5 kHz. Energy around 1.2 kHz may also have been partly responsible for male voices sounding "bright" and "forward" to some panel members.

Figure 8 shows the left-channel versus right-channel output when playing a recording of pink noise. A perfect match between channels would result in a 45° straight line. Figure 9, which shows the phase versus frequency response for the same recording, indicates that the interchannel phase difference occurs mainly above 5 kHz. This interchannel difference appears to be trivial: The panel members rated imaging performance the same for the tested arm/cartridge combination and the reference system.

The ability of the WT/vdH combination to track high-level middle-register signals (Fig. 10) is very good, if not exceptional. The 19.2-cm/S, 1-kHz signal is very difficult to track, and the performance of this combination puts it in the company of some of the best arm/cartridge pairings. The spectrum produced by this tracking test (Fig. 11) indicates that the third and fifth harmonics, especially in the right channel, will cause the sound to be perceived as being a bit bright, which correlates well with most of the comments from the listening panel.

Figure 12 shows the output due to a 10.8-kHz tone burst. There is a little compression at the top of the 30-cm/S burst which could affect the sound of high-level high frequencies. In this regard, however, the rating and comments of panel members about the sound of cymbals, for instance, put the WT/vdH slightly above the reference system. The spectrum produced by the 10.8-kHz tone burst (Fig. 13) is a very good indication that this arm/cartridge combination does not produce a lot of low-frequency modulation garbage which can cloud the sound of loud passages that include high-level, high-frequency sounds.

Figure 14 shows the output produced when playing a 1-kHz square wave The rise-time of the van den Hul cartridge is extremely fast but well damped. This damping also correlates well with the amplitude versus frequency response (Fig 2), which shows a gentle roll-off above 20 kHz. I thought it might be interesting to see the spectrum produced by this 1-kHz square wave (hence Fig. 15) Calculating such spectra for each tonearm/cartridge combination might prove valuable for reference and to indicate their tonal balance.

Conclusions

At the end of a long and detailed discussion of the measurements and sound quality of a combination such as this, I am supposed to say something pithy and concise so you won't have to read through all the details. It isn't easy in this case. Some of the panel members liked the WT/vdH very much, and I must admit that, on a lot of program material, it was very precise, dry, and analytical. It was judged slightly better than the reference system on strings, guitar, and piano, as well as for general clarity; it was judged not quite as good on voice, bass, rock, and for spaciousness The two systems tied on brass drums, and full orchestra, as well as image stability

You can't accidently damage the stylus by dropping the tonearm because it takes about 1 S to fall from a horizontal starting position The Well Tempered arm can tame cartridges that require good damping The height of the arm base, however, may cause problems with some turntables unless the dust cover is removed.

I found the high-frequency tracing capability of the van den Hul stylus to be excellent, and I never lost my temper while using the tonearm. For the rest of the conclusions and comments of the listening panel, you will have to read the report.

Edward M. Long

The suspension of the Well Tempered turntable reduces motor and platter vibrations more than it does external ones.

Continued from page 110

Firebaugh decided to take a different approach. The Well Tempered turntable has what I would call moderate mass, but it has no springs. Instead, it sits on four rubber feet which are fastened to the bottom with screws. Rather than fight the battle of trying to tame the tendency of suspended turntables to rock up and down, side to side, or both, Firebaugh decided to tackle the problems caused by bad-tempered vibrations above the seismic region. He has applied his "well tempered" viscous damping techniques to the motor bearing and the main turntable platter bearing.

The motor is a 24-pole a.c. type that is highly modified. It has a new thrust bearing at the bottom, upon which the motor shaft rests. The bottom of the motor is mounted in a heavy, lead block which Firebaugh refers to as a "brick." Sometime back, there was a flurry in audiophile circles about the taming characteristics of bricks (of this or that material) when placed on amplifiers, CD players, etc. Perhaps that is the connection, but this brick is really beneficial since it gives tremendous stability to the motor.

Stability is very important in this case, because the motor of the Well Tempered turntable is completely separate from the rest of the machine and is connected to it only through the belt; it needs the mass to remain in position. The shaft of the motor turns in a bath of viscous fluid which smooths the rotation and reduces the tendency of the a.c. motor to "cog." The turntable platter shaft also runs in a bath of viscous fluid, and this tempers the rotation of the platter and helps reduce flutter.

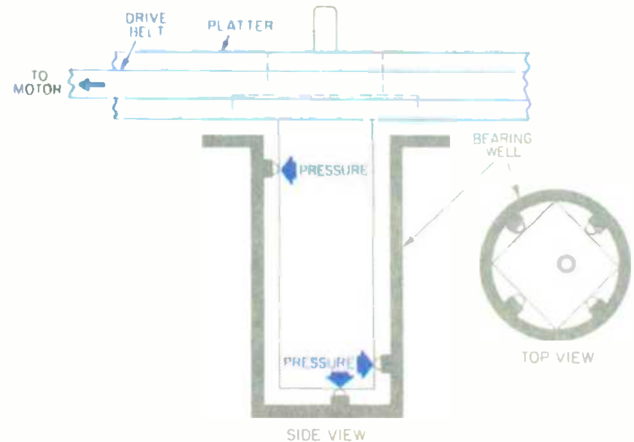


Fig. 1—Cross section and top views of the turntable shaft bearing. Tension on the drive belt presses the thick shaft against the thrust bearings on the sides of the shaft well.

First Impressions

The translucent white platter sitting on the three-tiered, satin-black base certainly gives the Well Tempered turntable a striking appearance. The turntable and base are one piece, unlike most other designs, and I must admit that I wondered how it would do during the usual shock tests. When I saw that the motor was completely separate, I remembered having seen the same technique on some early disc-cutting turntables. This approach was taken because some of the early, high-torque motors required to turn the massive cutting turntables had excessive vibration. Good ideas of the past have a way of returning, and, although this motor has very little vibration, separate mounting has to help.

Mounting a tonearm—other than the Well Tempered arm—to this turntable will not be a trivial task. There is no separate mounting board, and the base is solid in this area, except for a hole precut to fit the Well Tempered tonearm. The turntable cannot be ordered precut for other arms (Firebaugh has not yet tested the table with any), and I am certain that changing to a different tonearm later would not be easy.

I also noticed that the drive belt seemed thinner and narrower than most of the other belts I have seen. Firebaugh recommends putting a single twist in the drive belt, between the motor pulley and the turntable platter, and this looks a little strange at first.

MEASURED DATA

PARAMETER	MEASURED*	COMMENT
Speed Accuracy	0.3% fast	
Speed Stability	±0.21%	Very good
Wow, DIN Unwtd.	0.16%	Very good
Wow, DIN Wtd.	0.12%	Good
Flutter, DIN Unwtd.	0.13%	Very good
Flutter, DIN Wtd.	0.02%	Excellent
Wow & Flutter, DIN Unwtd.	0.24%	Very good
Wow & Flutter, DIN Wtd.	0.14%	Good
Short-Term Drift	±0.15%	Very good
Rumble, Unwtd.	−66.8 dB	Excellent
Rumble, Wtd.	−81.8 dB	Excellent
Suspension Resonance	Not applicable	No suspension

*Measured with one twist in drive belt; see text.

Features

The base of the Well Tempered turntable is part of the turntable and not a separate entity, as it is on most other turntables. It measures 19 inches wide, 15⅜ inches deep, and 5¼ inches from the bottom of the rubber feet to the top of the center spindle, which is the highest point. When the record clamp is in place, the total height is about 5¾ inches. With the Well Tempered tonearm, which is higher than most, the height with the clamp is 7½ inches. The table weighs 26 pounds without the motor, which is completely separate. The base is solid and consists of three pieces of medium-density fiberboard, each 1 inch thick, plus two

Instead of riding in the usual sleeve bearing, the shaft is pulled by the belt against diagonally opposed thrust-bearing pads.

layers of damping material between them. The rounded edges on the base, and the satin-black finish, give it a very nice appearance. A 5¼ × 5¼-inch cutout on the left side of the base allows for the motor, which sits in this area. Four soft rubber feet are fastened to corners of the lead brick to which the motor is mounted.

The main bearing well is made of black Delrin. The body of the well is 2.89 inches deep and 1.492 inches in diameter. It has a collar, 1.75 inches in diameter and 0.10 inch thick, with a "V" notch that must be pointed toward the motor when it is installed in the mating hole in the turntable base. Instead of the usual sleeve bearing in the spindle well, there are two Teflon thrust-bearing pads at the top, on the side facing the motor, and two near the bottom on the side away from the motor (Fig. 1). The drive belt pulls the spindle shaft against them! It is a weird experience to have the turntable platter move down against the base when pressure from a record cleaning brush, for instance, is applied on the right side. The one-piece record spindle/platter shaft is pressed into a hole in the center of the main platter. It has a half-inch-diameter shaft with a flat bottom which turns on a bearing at the bottom of the main bearing well. The upper part, which fits into the platter, is 0.725 inch long and 0.75 inch in diameter. It has a 0.980 × 0.125-inch lip that carries the weight of the platter and keeps the spindle shaft in place. The top section of the spindle has a diameter of 0.287 inch to fit the record hole.

The 11.5-inch diameter turntable platter is made of solid, translucent acrylic and has a 0.75-inch diameter hole in the center for the metal spindle shaft. There is a recess, 4.085 inches in diameter and 0.050 inch deep, to accommodate the extra thickness of a record at the label area. The platter is 0.915 inch thick at the edge and is machined in such a way that the surface is dished down toward the center to allow an LP to be pulled down by the record hold-down clamp. The clamp is 1 inch high and 1.75 inches in diameter, with two knurled rings for better gripping. The design of the record clamp is very clever, although it can be used only with the Well Tempered turntable. Inside its center hole is a recessed #10-32 hex-head bolt which mates with #10-32 threads inside the spindle. When you turn the clamp on the spindle, it rotates down and pushes the record against the turntable, making it easy to adjust the force to the amount desired.

The separate 24-pole a.c. motor weighs about 2 pounds. It has been highly modified with a new thrust plate and the addition of viscous fluid inside the motor bearing. The motor mounts to a 0.25-inch thick, 4.5 × 4.5-inch soft iron plate that provides excellent shielding of the motor's magnetic field. The motor is mounted on the 10-pound lead brick and is sandwiched between the steel top plate and the brick by two long screws. A pushbutton a.c. power switch is mounted on the top plate. The 5-foot a.c. power cord exits from the bottom of the brick. The motor pulley, which is force-fitted to the motor shaft, is made of Delrin and has two steps, for 33⅓ and 45 rpm. Speed change is accomplished manually by moving the belt to the proper step on the pulley. The motor has good torque and accelerates the turntable from zero to full speed in 1.5 S. When the power is turned off, the turntable platter comes to a full stop in 2.5 S. The drive belt

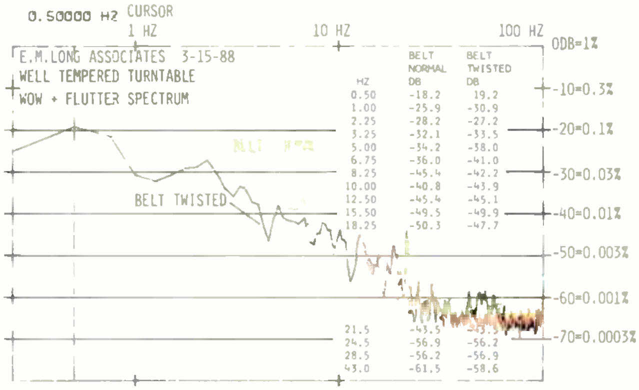


Fig. 2—Wow and flutter spectrum, from 0 to 100 Hz. Note the improvement in performance with the belt twisted (see text). The arm cartridge resonance with the van den Hul MC-One cartridge and Well Tempered tonearm is at 10 Hz. This resonance, which is normally visible in such plots, cannot be seen due to the arm's high damping.

Fig. 3—Speed drift over 42-S period, with no twist in belt. Cyclic variations at 1.8 S, or 0.56 Hz, are related to the rotational speed.

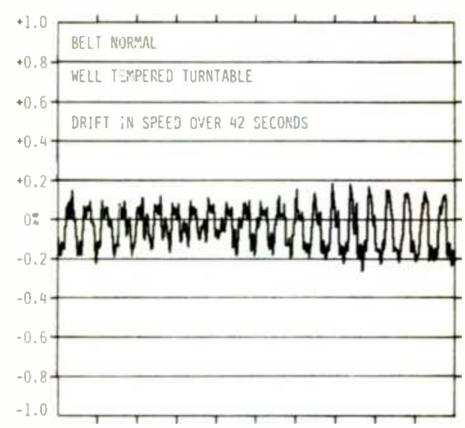
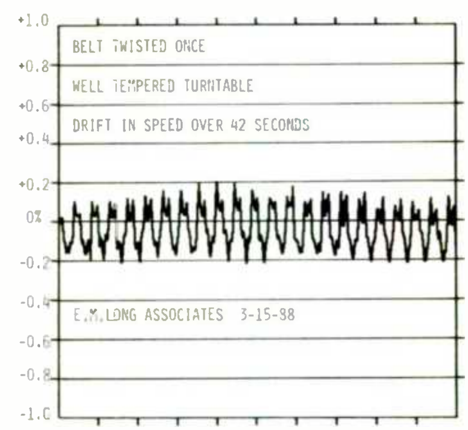


Fig. 4—Same as Fig. 3 but with belt twisted.



The separate motor housing is a revival of a good idea from the past, when it was used on disc cutters.

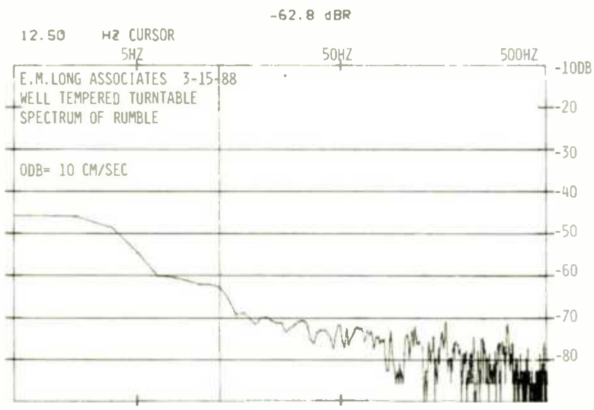


Fig. 5—Rumble spectrum. Due to the well-damped arm resonance, the output is extremely low.

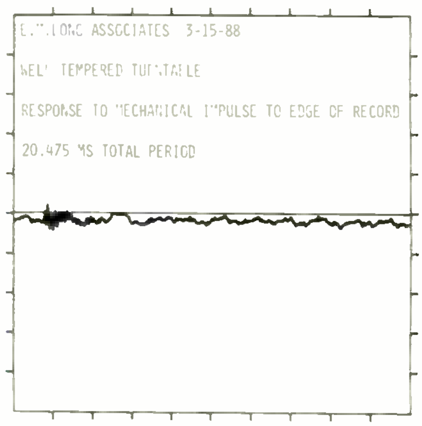


Fig. 6—Output vs. time for mechanical shock applied to edge of a stationary record, with stylus resting in groove.

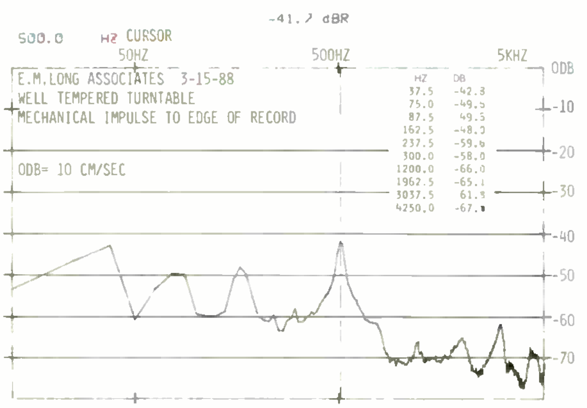


Fig. 7—Spectrum to 5 kHz of output from a series of 16 mechanical impulses (averaged) applied to edge of a stationary record, with stylus resting in groove.

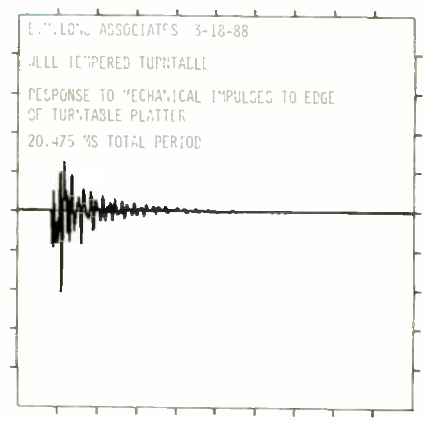


Fig. 8—Output vs. time for mechanical impulse applied to edge of the turntable platter, measured with accelerometer on the platter's edge.

is thinner than usual, only 0.0135 inch thick and is 0.125 inch wide. It wraps around the outside edge of the platter and the motor pulley.

Measurements and Listening Tests

Bill Firebaugh tells me that, during the design and testing of the Well Tempered turntable, one time he accidentally put a single twist in the drive belt while putting it over the motor drive pulley. He didn't notice this right away and continued testing. The wow and flutter measurements were lower than usual, and when he looked at the turntable, he saw the twisted drive belt. Firebaugh reasoned that perhaps the twist in the belt made it come off and go back on the platter more smoothly as the platter rotated. It seemed to him that the reproduced sound from records, particularly piano music, was more realistic. To check this out, I listened to some piano music with and without a twist in the drive belt. I think Firebaugh is right; I found that the piano sounded better with one twist in the belt. (I don't know if this would work as well with other belt-drive turntables because I haven't had the chance to try it.)

Figure 2 shows the wow and flutter spectrum with and without the belt twisted. The level below about 10 Hz is lower with the belt twisted, except at 2.25 Hz. There is more output in the range from around 50 to 60 Hz, but it is extremely low in both cases, and the flutter readings listed

Twisting the drive belt may seem to be a strange idea, but it did improve both the sound quality and the measured performance.

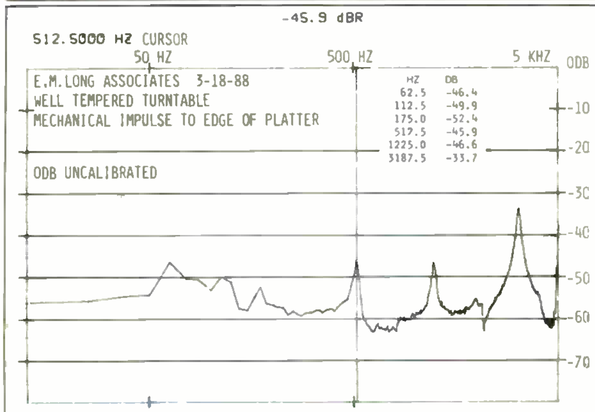


Fig. 9—Spectrum to 5 kHz of the vibrations caused by mechanical impulse applied to edge of the turntable platter (16 impulses, averaged).

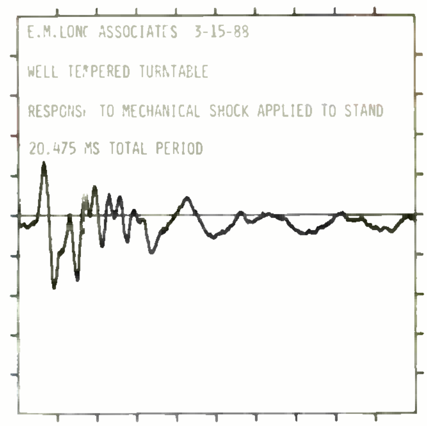


Fig. 10—Output vs. time for mechanical shock applied to heavy stand on which the turntable rested.

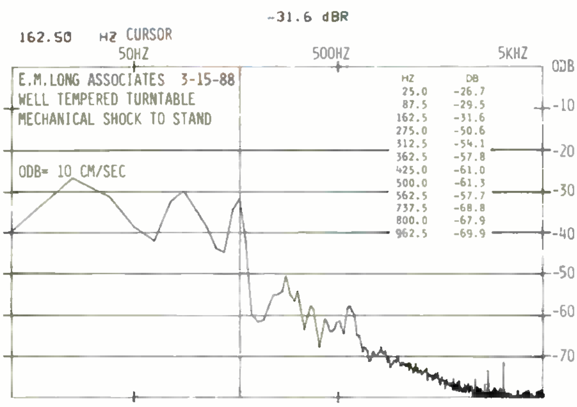


Fig. 11—Spectrum to 5 kHz of the vibrations caused by mechanical shock applied to turntable stand (16 impulses, averaged).

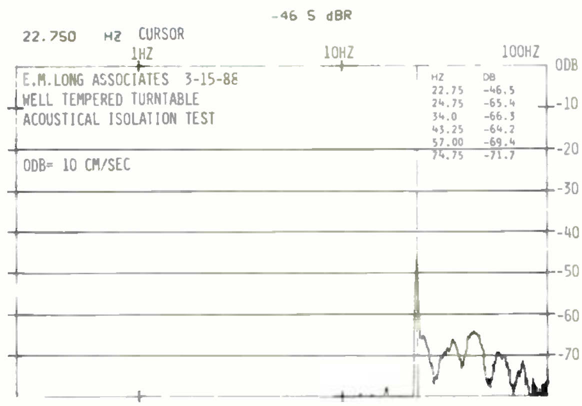


Fig. 12—Spectrum to 100 Hz of the vibrations from an acoustic field of 100 dB SPL at the surface of a record. Stylus is resting in a groove near the middle of the record. Acoustic isolation is excellent.

in the Measured Data table, which would be affected by the output in this range, are excellent. During the listening evaluation, I put a twist in the drive belt and left it that way. Comments from members of the listening panel indicate that they thought that the sound of the piano was a little better with the Well Tempered turntable than with the reference system: "Sustained chords were very steady" and staccato passages "very precise."

With the belt normal, the variation in rotational speed for 33⅓ rpm (Fig. 3) looks a little strange, since it is not consistent. However, if you compare it with data from reports I have done on other turntables, you will see that the greatest and smallest variations in speed on the Well Tempered turntable are comparable to results from some of the best units tested. Figure 4 shows that the speed also varies with the belt twisted and that there is a smaller variation over the 42-S period measured. That, too, is comparable to some of the best tables.

The rumble spectrum (Fig. 5) shows something I have tried to emphasize in past reports: That the tonearm/cartridge resonance is one of the major contributors to rumble as well as to wow and flutter. The mass of the Well Tempered tonearm, and the compliance of the van den Hul MC-One cartridge used during my evaluation of the Well Tempered turntable, resonate at 10 Hz, but the Q of this resonance is very low, due to the arm's excellent damping. In

The wow & flutter and rumble spectra show how turntable performance gains from well-damped arm/cartridge resonance.

other words, when this resonance is excited by rumble or wow, there is very little output. Figure 5 shows very little rise in output around 10 Hz because of this. Look at Fig. 2 again and you will see that, at 10 Hz, there is also very little indication of the tonearm/cartridge resonance. This means that if I took only readings from a meter while testing for rumble or wow, I would get lower readings when the tonearm/cartridge resonance is well damped (as it is in this case) than I would if I used a poorly damped tonearm/cartridge combination. The turntable would be rated better or worse depending upon the arm/cartridge combination used to test it. That is why I show the *spectrum* of the rumble and the wow and flutter. Although the tonearm/cartridge resonance must be considered as affecting the overall quality of the sound, when trying to rate the turntable by itself—with respect to rumble or wow—the amount of output around the tonearm/cartridge resonance should be ignored. I purposely played a recording that always seemed to have more than a normal amount of rumble and asked the panel to comment on the background rumble. Their response was that the Well Tempered turntable, with the Well Tempered tonearm/van den Hul cartridge combination, produced a little less rumble than the reference system, whose tonearm/cartridge resonance is less well damped.

Figure 6 shows the output produced when a mechanical impulse was applied to the edge of a stationary record, with the stylus of the van den Hul cartridge resting in a groove near the middle of the record. The output is very low, and since it dies out quickly, it is also very well damped. Figure 7 shows the spectrum of the output caused by 16 mechanical impulses. The main output is at 500 Hz; this could affect the sound, especially that of voices. During the listening evaluation, some panel members said the sound was "bright" and "forward" for the Well Tempered turntable and "clearer" for the reference system when reproducing voices. Energy peaks at 1200, 1962.5, 3037.5, and 4250 Hz might also be related to the rather consistent comments about the Well Tempered turntable being "bright" on most program material. I wanted to determine whether the energy around 500 Hz was due to the resonance of the acrylic turntable platter, so I made a large number of tests using an accelerometer. The result of one of these tests is shown in Figs. 8 and 9. The output versus time for a mechanical impulse applied to the edge of the platter is shown in Fig. 8; the result is similar to that seen in Fig. 6, albeit at a higher level. The level of the tests with the accelerometer is uncalibrated, so only the shape of the envelope and the spacing of the up-and-down undulations is of interest. Comparing the spectrum shown in Fig. 9 to that in Fig. 7 will yield some useful insights about the resonant modes of the platter. Peaks of energy around 500, 1200, 2000, and 3000 Hz can be seen in both figures. When a record is clamped to the Well Tempered turntable, the output at all but the 500-Hz peak is reduced considerably, but some energy is clearly finding its way into the reproduced sound.

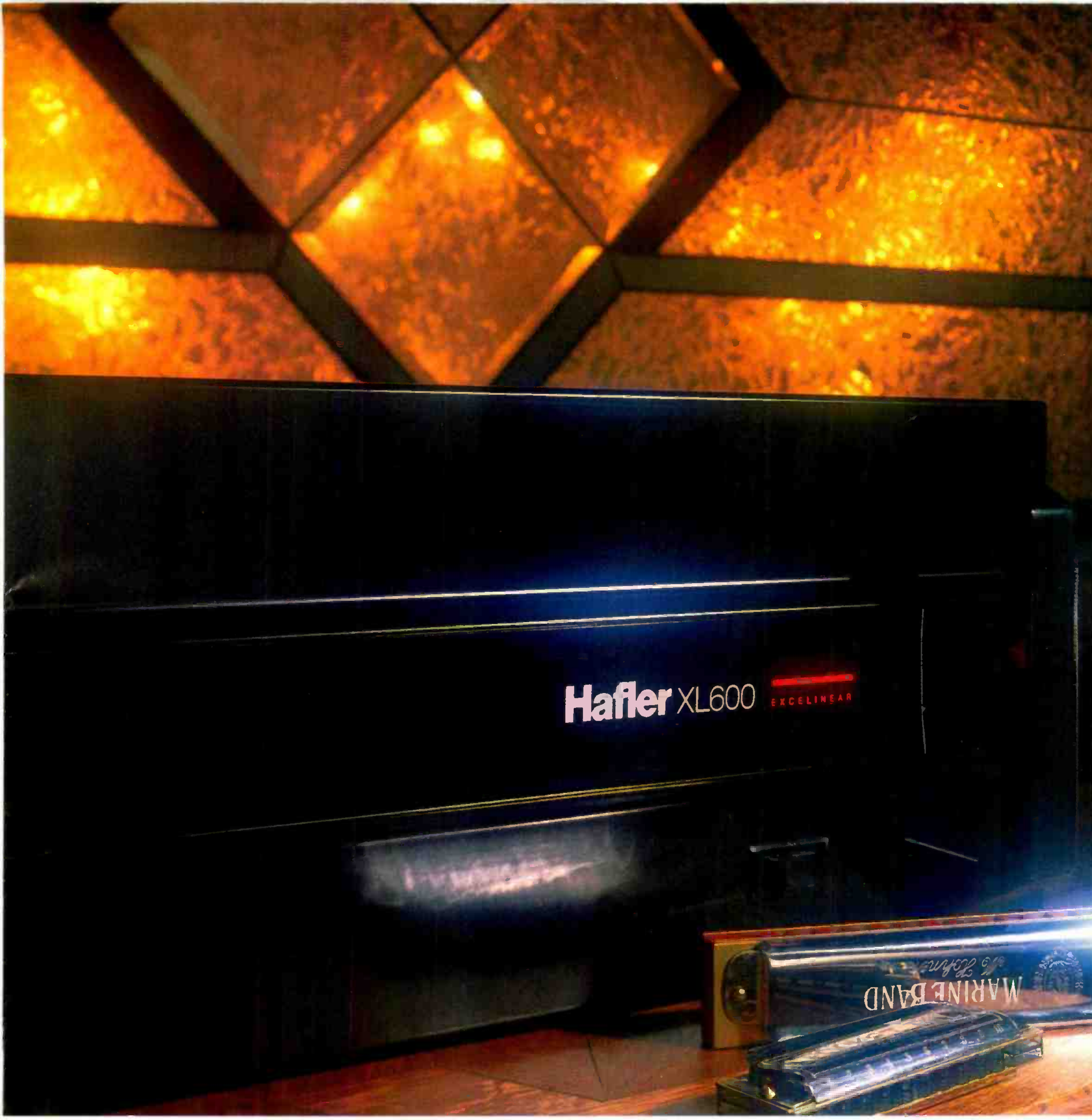
Figure 10 shows the output over 2.05 S for the Well Tempered turntable when a mechanical shock was applied to the heavy test stand upon which it was resting. The results are good, considering that there is no suspension and that the turntable has only four rather ordinary rubber feet. I think that excellent damping of the Well Tempered tonearm/van del Hul cartridge resonance is responsible for this. If a different tonearm/cartridge combination were to be used, the results might well be worse. The spectrum of the output due to 16 mechanical shocks applied to the test stand is shown in Fig. 11. The greatest output is in the range below about 170 Hz, with major peaks at 25, 87.5, and 162.5 Hz.

Figure 12 shows the spectrum of the output due to a 100-dB SPL acoustic field at the surface of a record. The stylus was resting in a stationary groove while the acoustic signal was slowly swept from 20 to 100 Hz. The output is very low, which indicates that acoustically induced feedback should not be a problem.

Conclusions

I must admit that, although I hope you will read my entire report on the Well Tempered turntable, this is the part I usually jump to when I read other reviews. The Well Tempered turntable produced excellent results in both the technical measurements and the listening sessions. Some panel members, but not all, preferred the sound of the Well Tempered system (which included the Well Tempered tonearm and the van den Hul MC-One cartridge) to that of the reference system. For those characteristics of a turntable which can affect the sound quality, without direct reference to the tonearm/cartridge combination, the Well Tempered turntable did very well. The pitch is very steady, the torque is high enough so that the speed remains constant during soft and loud passages, and the immunity to mechanical shock and acoustical feedback is very high. The Well Tempered turntable's unusual design, with the motor separate from the main body of the turntable, provides excellent isolation of mechanically transmitted vibrations. However, it is also something to consider when thinking about mounting or transporting the turntable. Such innovative design ideas may raise questions in your mind, as they did in mine, but they seem to work very well. If you are after a high level of sound quality, you should check out the Well Tempered turntable. I think you will like it.

Edward M. Long



Hafler XL600
EXCELINEAR
MARINE BAND

16

REVOX B215 CASSETTE DECK

Manufacturer's Specifications

Frequency Response: 30 Hz to 18 kHz; to 20 kHz with CrO_2 and metal tapes.

Harmonic Distortion: 0.8%.

Signal/Noise Ratio: 72 dBA with Dolby C NR.

Separation: 40 dB.

Erasure: 70 dB.

Input Sensitivity: 50 mV.

Output Level: Line, 775 mV; headphone, 2.8 V.

Flutter: ±0.1% wtd. peak.

Fast-Wind Time: 75 S for C-90 cassette.

Dimensions: 17.7 in. W × 6 in. H × 13.1 in. D (45 cm × 15 cm × 33 cm).

Weight: 20.4 lbs. (9.2 kg).

Price: $2,300; with black and gold front panel, Model B215-S, $2,800; Model B208 romote-control transmitter, $160.

Company Address: 1425 Elm Hill Pike, Nashville, Tenn. 37210. (Originally published July 1985)

Photograph: ©Bill Kouirinis

The Revox B215 cassette deck uses sophisticated microprocessing for many internal functions. There are actually three microprocessors: One for the time counter, another for the automatic tape-matching system, and the third for housekeeping and for control interfacing with other components in Revox's B200 series.

All units in this series can be operated from the same optional remote-control unit. But they can also connect, via rear-panel serial ports, to a separate interface box which can then be connected to a home computer (for programmed control) or to infrared remote-control receivers in other rooms. With the Revox units interconnected in this way, one could simultaneously start the B215 tape deck and switch the receiver to "Tape" mode by pressing "Play" on the remote transmitter—whether the transmitter is pointed at the receiver, the B215, the interface unit, or an infrared receiver in another room.

The most important use for the on-board microprocessing is the automatic alignment to match the characteristics of any tape used. In just 20 S, adjustments are made automatically to bias, record sensitivity and equalization to ensure flat response, good Dolby NR tracking and low distortion. Information on the internal settings can be stored for two Type I tapes, three Type IIs, and one Type IV. The B215 also incorporates the Dolby HX Professional system, which varies bias during recording in accordance with the spectral makeup of the signal for lowest distortion overall.

The microprocessor-controlled time counter yields elapsed-time indications after only a few seconds of play, no matter where the cassette is started. A selected elapsed time can be entered, and a fast-wind made to that point. Two time addresses can be stored for one-button fast-wind returns, or for looping (continuous play) between them.

Another helpful feature of the B215 is a system which automatically sets recording levels. Automatic fade-in and fade-out during recording is an additional nicety.

The tape drive uses four motors, two for the direct-drive capstans and two for spooling the tape. An optical end-of-tape sensor stops the transport at the start of the clear leader, instead of at its end. This positions the tape exactly where recording can be restarted as soon as the cassette is flipped; time is not lost while the leader passes the heads once in each direction.

Control Layout

The B215 deck is large, but it has a friendly look, with brushed aluminum as the top of the front panel and dark gray for the lower part. The black designations on top and the white ones below are very easy to read over a wide range of lighting levels, making the B215 one of the best units in this regard. The very large, aluminum pushbuttons and the large, medium-gray and red ones all stand out clearly from the panel and require just a light touch for actuation.

After the deck is plugged in but before it is turned on, a red standby indicator illuminates in the "IR-Sensor" window at the upper left end of the gray panel. The deck can be turned on in either of two ways, with the B205 remote control or with the "Power" pushbutton at the upper right of the front panel. With turn-on, the red indicator goes off, and the "Real Time Counter" and "Peak Program Indicator" LCDs appear. The counter display shows "--:--" over "Min" and "Sec" to remind the user that calibration has not been done for an elapsed-time indication. The "Peak Program Indicator" has "L" and "R" horizontal meter scales and calibrations from "−30" to "+8" in between. Just to the right of the meters is "Bal" with an arrow above it, pointing up (next to the "L" scale), and an arrow below it, pointing down (next to the "R" scale). At the lower left of the same display area, "Source" announces that the incoming signal is being monitored. Additional details of these displays will be given while discussing the use of the pushbuttons.

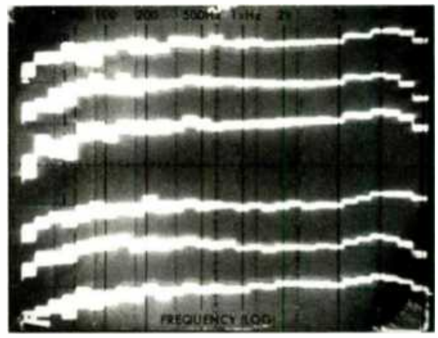


Fig. 1—Record/playback responses using Dolby C NR. Top three traces made with Maxell UD-XL I (Type I), TDK HX-S (Type II) and TDK MA-R (Type IV), all at Dolby level. Bottom three traces with the same tapes but at −20 dB. (Scale: 5 dB/div.)

To the right of the displays are the "Set Level" and "Fade In/Out" pushbuttons. "Set Level" automatically sets the digitally controlled input-level attenuator while you play the loudest portion of a disc, so that the highest recording levels will be just below the point where unacceptable distortion would occur. Automatic setting continues as long as the button is held in, so the actual time taken is determined by the user.

With the "Fade In/Out" button, the signal can be faded between full off and the preset attenuator level, whenever desired. You cannot, however, vary the fade speed or interrupt the fade halfway. Fades can be made any time during recording without stopping the transport.

Fading is also invoked by the "Pause" control, which is grouped, with the other transport-control buttons, to the fade button's right. There is an automatic fade-in if recording is started from record-pause mode (rather than "Stop"), and an automatic fade-out if you interrupt recording with the "Pause" instead of the "Stop" button. Pressing "Pause" also automatically switches the monitoring back to "Source," in anticipation of continued recording—a convenient feature.

It is possible to switch among modes as desired, and punch-in recording is possible by holding "Rec" and "Play"

A helpful feature automatically sets the recording level so the highest peaks are just below the distortion point.

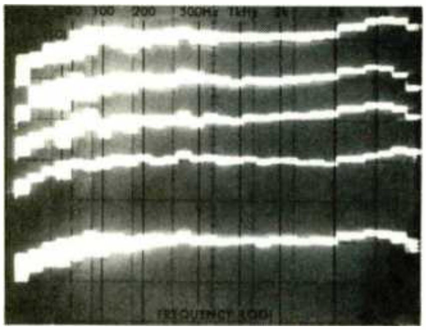


Fig. 2—Record/playback responses. Upper four traces, all made with Dolby C NR, are: +6 dB on Maxell UD-XL I, +4 dB on TDK HX-S, +6 dB on TDK MA-R, and +10 dB on Maxell UD-XL I. Bottom trace shows overlaid responses with Dolby B and C NR and without NR, all made on UD-XL I tape at −14 dB. (Scale: 5 dB/div.)

at the same time. The above constitute a nice collection of features for the serious recordist.

In "Rec/Pause," the meter display shows "Source" and, above, a flashing "Record." Pushing "Pause" again initiates recording, with the display indicating the change in monitor status from "Source" to "Tape."

Below the transport buttons are nine gray pushbuttons plus the "Store" button, which is red. The top row consists of "Loop," "Recall," and two "Address" buttons, "Loc 1" and "Loc 2." The next row is for "Cancel" and the aforementioned "Store." The bottom row has "Save Status," "Play Time," "Min," and "Sec."

When a cassette is first inserted, "Real Time Counter" is blank, as mentioned earlier. With a push of "Play Time," a standard tape length (whichever you used last) will be displayed; successive pushes will step the indicated length from "C 46" to "C 60" to "C 90" to "C 120," and back to the start again. After the selection of the correct length, a few seconds of playing or recording will get a calibrated, elapsed-time reading in the counter display. After calibration has been completed, a start of recording will automatically store the "Min/Sec" address (tape location) in "Loc 1." By use of the "Min," "Sec" and "Store" buttons, and then "Loc 1" or "Loc 2," any location on the tape can be put in memory. Except when in record mode, a push of "Loc 1" or "Loc 2" will initiate a fast-wind to that exact point on the tape. The counter display shows "Loc" and "1" and/or "2" above it when there is an entry or two to indicate. When both locations are used, a push of "Loop" will initiate continuous play and rewind cycling between the two points, even fast-winding to the start point from any location on the tape. Arrows appear between the tops and bottoms of the "1" and "2" in the display, reminding the user that the deck is in "Loop" mode. "Recall" and a location button will get a display of the corresponding tape-time location. "Cancel" will, of course, clear the memory of whichever button is pushed.

"Save Status" is used to store all recorder settings including level, NR system, balance, etc., in a nonvolatile memory for use with a timer (which, of course, shuts off all power to the recorder for a period of time).

Under the counter and meter displays are 11 pushbuttons, 10 gray and one red. The top row, just to the right of the infrared sensor mentioned earlier, has two buttons for input level ("−" and "+") and two for balance ("L" and "R"). When an input-level button is held in, a relative level from "−∞" to "+10" appears in place of the "Min/Sec" readout. A brief push will get single steps up or down, and a hold will obtain continuous stepping which increases in speed as the button is held in. The arrows above and below "Bal" show when there is electrical balance, but the level indication must be used to find the best setting.

The second row of buttons under the displays consists of "Tape Type," "NR System," "MPX," and the red "Align." When a cassette is first inserted into the holder, tape type is automatically sensed and displayed, provided that the cassette has the sensing holes which indicate this information. "Tape Type" allows manual setting for "Type I," "Type II," "Type II—120 µS," and "Type IV." All are self-explanatory with the exception of "Type II—120 µS." This is an unusual and useful feature for the serious recordist: If there is a more-than-average amount of energy in the higher frequencies, the results with a Type II tape may be better with 120-µS EQ instead of the usual 70 µS. The selected tape type is announced along the bottom of the meter display, Type I to Type IV, left to right.

The selection of "MPX," "Dolby B," or "Dolby C" is similarly indicated along the top of the meter display. "MPX" is an on/off selection, and "NR System" steps the choice from off to "Dolby B" to "Dolby C" NR.

Alignment, in the case of the Revox B215, means electronic adjustment of the recording function and not the mechanical adjustment of a record or playback head. A push of "Align" with the deck in record/pause mode starts the process that adjusts bias, record sensitivity and equalization for the best responses with low distortion, both with and without Dolby NR. It's a 20-S procedure, and while it's functioning, "Align" appears at the lower right in the meter display. There are a total of six memory locations for alignment information: Two for Type I tapes (A1 and A2), three for Type II (A1, A2, A3), and one for Type IV (A1). With the use of "Align," the settings are automatically put into memory, normally A1 location. To save the settings for another tape formulation without disturbing the information in memory A1, push "Align" and then the "Pause" button to step to the next memory location. Overall, this is a very good way to handle tape matching, with the convenience of storing the matching-condition information for the tapes most used. These

The drive, which had a look of long-term durability, ran very quietly—perhaps the best of any I've yet tested.

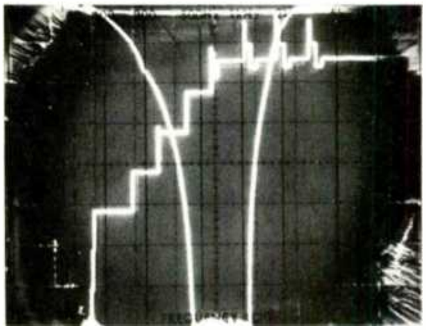


Fig. 3—Tests of two gain-adjustment functions. Curved trace shows fade-out from 0 dB to maximum attenuation, and fade-in from maximum attenuation to 0 dB. Stepped trace shows action of "Set Level" function as input is increased in 10-dB steps from −70 to 0 dB (see text). (Vertical scale: 10 dB/div.; horizontal scales, 1 S/div. for fader, 5 S/div. for "Set Level" test.)

memories are also nonvolatile, holding their contents even if the recorder's power is disconnected.

Along the bottom row of buttons below the display are the headphone jack, all the way to the left, two "Phones Volume" buttons ("−/+") and the "Monitor" selector (causing "Source" or "Tape" to appear in the meter display). The headphone level can be set to one of eight steps. My immediate reaction to this design was a bit of skepticism, but I reserved judgment until I actually tried listening.

The shallow, vertical well for the cassette has a very open design, which gives outstanding access for any sort of cleaning or demagnetizing. Inserting a cassette was a simple process of putting the top in first, then pushing in the bottom. I liked the finish and the ruggedness of the drive elements, particularly the large diameter of the capstan shafts.

On the B215's back panel are the expected stereo pairs of in/out phono jacks. There is also a DIN-type socket for the serial interconnection link with other Revox equipment. The power cord is detachable.

Removing the steel top and back covers allowed examination of the interior. The chassis has a rigid, box-girder construction, providing excellent support for the transport system and the circuit cards. The large flywheels were very evident, and the rest of the drive was judged to be very well-constructed, with a definite look of long-term reliability. The drive was very quiet, even in play mode—perhaps the quietest of any deck I've tested to date. The soldering on the cards was excellent, with slight flux at a few hand-soldered points. There were a total of four fuses, all in clips.

Measurements

The playback responses of the Revox B215 were the best I have measured to date, with many points within ±0.3 dB of the reference level. Playback of a standard flux level was indicated correctly, and tape play speed was 0.2% fast, at the most.

For record/playback measurements, I used "Align" to match the deck to a large number of tapes having a wide range of bias and sensitivity characteristics. For the test signal, I used what I call "PN/Music"—pink noise rolled off at 2 kHz—to ensure accurate assessment of the performance with Dolby C NR. (Testing with sine-wave signals can give a misleading impression of response irregularities with Dolby C NR.) The record/playback responses were at least very good with *every* tape tried, and excellent with most. Maxell UD-XL I, and TDK HX-S and MA-R, were judged to be the best overall and were therefore used for the detailed tests that followed. Excellent results were also obtained with these Type I tapes: BASF Pro I Super, Fuji FR-I, Maxell XL I-S, PDMagnetics Tri-Oxide Ferro HG, Sony AHF, TDK AD and AD-X, and Yamaha NR-X. Type II tapes with excellent results included BASF Pro II Chrome, Fuji FR-II, Maxell UD-XL II and XL II-S, PDMagnetics 500 Crolyn HG, Sony UCX and UCX-S, TDK SA-X, and Yamaha CR. Among Type IV tapes, Maxell MX, Memorex Metal IV, Sony Metallic, TDK MA, and Yamaha MR were excellent. I was further impressed by the fact that the B215 got very good responses with BASF Metal IV in the C-120 length, much better than other decks I have tried.

Revox did not provide detailed information on the alignment process, but a little detective work with the aid of my Hewlett-Packard computing counter got these clues: There is a sequence of four tones—17.4 kHz, 477 Hz, 17.4 kHz, and 3.7 kHz—with many stepped-level changes in the first three tones and a relatively small and smooth change in the level of the final tone. The deck's output was muted during "Align," but it was possible to observe the sequence with playback later. There were many changes during the 20-S process, and I could see that there were many comparisons made between 477-Hz and 17.4-kHz outputs at a number of absolute levels. It appeared more than likely that settings for bias and record sensitivity were very accurately set for good responses and low distortion. The 3.7-kHz level adjustment was judged to be the final touch-up for the flattest responses across the band.

Figure 1 shows record/playback responses, with Dolby C NR, for the three selected tapes, both at Dolby level and 20 dB below that. All of the responses are very flat, including those at 0 dB. (I should point out that with the PN/Music test signal, there will be less high-end roll-off in the playback because the rolled-off test signal causes much less tape saturation.) Having made that parenthetical note, I call attention to Table I, which lists the −3 dB limits for all three tapes, with and without Dolby C NR. These tests were made with sine-wave test tones which were *not* rolled off at the higher frequencies. The results were outstanding at Dolby level: The low-end responses dipped down 3 dB at 22 to 24

The B215 had the best playback responses I've yet measured. Record/play response was at least good with *every* tape, excellent with most.

Because it also controls other Revox components, the B208 remote unit has more buttons than the B215 tape deck.

Hz, came back up somewhat, and finally rolled off at 9.4 to 10.4 Hz. Figure 2 shows record/playback responses with PN/Music at higher levels. The outstanding Dolby NR tracking is illustrated in the bottom trace, where the results at –14 dB for no NR, Dolby B and Dolby C NR are all overlaid, making just one trace.

Table II lists a number of measured record/playback characteristics, all excellent. The measurement for 10-kHz phase error and jitter between channels was one of the best I have ever seen, and the multiplex filter was positioned exactly. The level of bias in the output during recording was very low.

The level of the third-harmonic distortion (HDL_3) was measured, with Dolby C NR, as a function of level for the three tapes, and as a function of frequency at –10 dB with TDK HX-S tape. Table III lists the distortion in the output from –10 dB to the points where HDL_3 equals 3%. The distortion-limit levels are somewhat low; but the distortion figures for 0 dB correspond very closely to specifications. The mid-band distortion was not as low as some other decks' (Table IV), but 0.24% at the frequency extremes is very good.

Signal/noise ratios were measured with and without Dolby C NR, with both IEC A and CCIR/ARM weightings. The results in Table V are a close match to other high-performance decks at Dolby level, but are somewhat low with reference to the 3% limit point. This does, of course, correlate to the somewhat low 3% points measured earlier. Perhaps we should recall the deck's outstanding frequency responses to remind us of the trade-offs involved in recorder design.

Table VI shows measurements obtained for a number of input/output properties. Everything seemed quite in order, but the overload level of 2.65 V calls for caution on the part of users who might feed the deck from equipment whose output capability is greater than this.

Figure 3 shows time and level plots of "Fade In/Out" and "Set Level." The sweep rate for the fades is 1 S/div. The fade-out takes about 2.4 S and the fade-in a little over 1 S, both acceptable times.

To test the "Set Level" function, I first attenuated the output from my test generator by 70 dB and set the B215's input level control to "–∞" to challenge the automatic function with an extremely low-level test signal. The B215 automatically and rapidly readjusted its input attenuator to its maximum setting, "+10," but, as Fig. 3 shows, the resulting record level was still only –35 dB.

I then increased the test generator's output in 10-dB steps. For the changes from –70 to –40 dB of generator output, the B215's attenuator remained at "+10," and recording level rose in accordance with the input-level changes. When the generator's output reached –30 dB, recorded level shot past its final limit, dropped briefly, then settled at the desired recording level, and the B215's attenuator reset itself to "+8." The action on subsequent 10-dB jumps in generator output was the same—a sharp rise, two sharp drops, and a final adjustment. The attenuator readout reflected these changes: "–2," "–12," and "–22." The B215 set its recording level below the distortion limit on these final four input-level steps. Because of its obvious stepping, "Set Level" should *not* be used during actual final recording, but it is a great convenience for setting up.

The input-level pushbuttons were used to make the deck's input attenuator step from maximum (+10) down to –∞. There were 1-dB steps from +10 to –44, followed by –46, –48, –51, –54, –60, and finally –∞. Each of the steps was substantially without error, and the tracking between sections was within 0.1 dB from +10 down to –54. These results are much superior to anything else that I have checked in the past.

There are eight positions for balance on either side of zero. The first "L" step, for example, increases "L" level by 1 dB; the second "L" step decreases the "R" level by 1 dB; the third step increases the "L" level by another dB, etc.,

Table I—Record/playback responses (–3 dB limits).

	With Dolby C NR				Without NR			
	Dolby Lvl		–20 dB		Dolby Lvl		–20 dB	
Tape Type	Hz	kHz	Hz	kHz	Hz	kHz	Hz	kHz
Maxell UD-XL I	22	21.1	8.2	23.1	23	14.1	8.5	24.6
TDK HX-S	22	22.7	8.3	24.5	23	16.0	8.6	25.5
TDK MA-R	24	23.4	8.4	23.1	24	17.0	8.8	23.9

Table II—Miscellaneous record/playback characteristics.

Erasure At 100 Hz	Sep. At 1 kHz	Crosstalk At 1 kHz	10-kHz A/B Phase Error	10-kHz A/B Phase Jitter	MPX Filter At 19.00 kHz
66 dB	59 dB	–93 dB	25°	7°	–32.7 dB

Table III—400-Hz HDL_3 (%) vs. output level (0 dB = 200 nWb/m).

		Output Level					
Tape Type	NR	–10	–8	–4	0	+4	HDL_3 = 3%
Maxell UD-XL I	Dolby C	0.10	0.14	0.27	1.0		+3.1 dB
TDK HX-S	Dolby C	0.14	0.22	0.46	1.1	2.7	+4.2 dB
TDK MA-R	Dolby C	0.17	0.22	0.50	1.2	3.0	+4.0 dB

Flutter was marvelously low, a consistent 0.10% weighted throughout a C-90 tape—the best I have measured to date.

until, with the eighth step, "L" has been increased by 4 dB and "R" has been decreased 4 dB. It is an interesting way of balancing, and it could be the best way, at that

The headphone volume adjustments were measured as: Maximum (0 dB), −4.1, −9.2, −14.2, −20.2, −28.1, −38.6 dB, and off. My first reaction was that the steps were too coarse, but trials revealed that the changes seemed quite right for whatever the user might desire—"a little softer," for example. I tried a number of headphones and found there was enough gain to drive any of them to very high levels.

Tracking between channels was outstanding so there was no need for balance trimming. The deck's output polarity was inverted in "Tape" but not in "Source" output mode.

Each of the horizontal bar-graph meter sections has 24 separate segments, although the bottom one in each meter is always on. Scaling extends from "−30" to "+8" with the lowest figures somewhat out of calibration. Accuracy was good from "−18" to "−6" however, and the single-dB steps from "−5" to "+8" were all within 0.1 dB—superb over this important recording-level range. The dynamic responses of the meters met the requirements of the standard for peak program meters with response to −1 dB with a 10-mS tone burst and a 1.4-S decay time. There were slightly higher meter indications with the tone-burst offset but there should have been more of a change. The frequency response of the meters was down 3 dB at 7.0 Hz and 169 kHz; the latter appears to be unnecessarily high.

Substantially no changes in tape play speed were detected with changes in line power from 110 to 130 V. Short-term variations in play speed were less than ±0.01%, excellent indeed. The flutter was marvelously low and very consistent throughout the length of a C-90, 0.010% wtd. rms and ±0.023% wtd. peak. After checking the effect of changing modes and loading and unloading the tape, I concluded that the B215 showed the best overall flutter performance I have measured to date.

The fast-wind speed was high, just 73 S for a C-90, but the stops were smooth and gentle. Times required for changing modes were very short, really too short to measure with a stopwatch. Cueing with fast-forward or rewind and "Stop" worked well, and seemed quite natural after a few trials. Calibration of the elapsed-time counter took about 7 S. With calibration made at the start of a cassette, errors built up during the playing, totalling a minute or so halfway through a C-90. Recalibration at that point reduced the error to several seconds, which is very acceptable. This is a good feature, but I would expect better accuracy. In case of any question, it would appear best to recalibrate the counter halfway through.

Use and Listening Tests

The owner's manual has a very good (albeit undetailed) text, well organized with helpful illustrations. Technical freaks would probably like more information on "Align" and the use of the microprocessors. The manual does not mention that punch-in recording is possible. Brief use pointed out to me that a cassette had to be advanced at least a short distance for "Align" to work, that was easy to do, and the benefits were great.

No record clicks could be detected by ear or meter, even when using Dolby C NR. There were very soft pause and stop "clunks" down in the tape noise (no indication on the monitoring meter). I found that with "Stop," and more so with "Pause," very short sections of the tape being used were not erased completely—leaving little beeps from my earlier tests. A very short rewind would be in order to prevent such distractions if a tape is being reused and has not been bulk erased.

For record/playback listening tests, I used pink noise for tracking tests and dbx-encoded disc versions of digitally recorded originals: *Wolftracks* with John Kay and Steppenwolf (Nautilus NR-53 dbx PS-1084), music of Rodrigo (Varese Sarabande VCDM 1000.150 dbx PS-1032), and others. The results were excellent, aided, I am sure, by the peak-responding meters, which were easy to read over a fairly wide range of illumination levels. With recording levels set quite high, I did prefer the Type II results over Type I, and the Type IV results over Type II. In each of these successive comparisons, the bass became less muddy and the music better detailed. Once again, I concluded that, with listening at high levels, the maximum recording level was best kept to that for a distortion of about 1%—about 0 dB on the B215.

The Revox B215 utilizes its microprocessors for many important and helpful things. "Align" performed very well, and the responses were among the best seen to date. Flutter performance was superlative, and the construction of the transport was judged to offer long-term reliability. The B215 is large, so it won't fit just anyplace, but it should have considerable attraction for those who seek performance and advanced features. *Howard A. Roberson*

Table IV—HDL$_3$ (%) vs. frequency at 10 dB below Dolby level.

		Frequency (Hz)						
Tape Type	NR	50	100	400	1k	2k	4k	6k
TDK HX-S	Dolby C	0.24	0.17	0.14	0.16	0.08	0.10	0.24

Table V—Signal/noise ratios with IEC A and CCIR/ARM weightings.

	IEC A Wtd. (dBA)				CCIR/ARM (dB)			
	W/Dolby C NR		Without NR		W/Dolby C NR		Without NR	
Tape Type	@ DL	HD=3%	@ DL	HD=3%	@ DL	HD=3%	@ DL	HD=3%
Maxell UD-XL I	67.5	70.6	52.0	55.1	68.6	71.9	49.4	52.5
TDK HX-S	69.0	73.2	53.[illegible]	57.3	69.8	74.0	50.6	54.8
TDK MA-R	69.1	73.1	53.3	57.3	69.9	73.9	50.7	54.7

Table VI—Input and output characteristics at 1 kHz.

Input	Level		Imp.,	Output	Level		Imp.,	Clip (Re:
	Sens.	Overload	Kilohms		Open Ckt.	Loaded	Ohms	Meter 0)
Line	47 mV	2.65 V	96	Line	779 mV	690 mV	1.5k	+16.0 dB
				Hdphn.	2.8 V	0.52 V	219	

17

NAKAMICHI CR-7A CASSETTE DECK

Manufacturer's Specifications

Frequency Response: 18 Hz to 21 kHz, ±3 dB.

Harmonic Distortion: 0.8%.

Signal/Noise Ratio: 66 dBA with Dolby B NR, 72 dBA with Dolby C NR.

Separation: 37 dB.

Crosstalk: −60 dB.

Erasure: 60 dB at 100 Hz.

Input Sensitivity: 50 mV.

Output Level: Line, 1.0 V; headphone, 12 mW into 8 ohms.

Flutter: 0.027% wtd. rms, ±0.048% wtd. peak.

Fast-Wind Time: 80 S for C-60 cassette.

Dimensions: $17\frac{1}{8}$ in. W × $5\frac{5}{16}$ in. H × 12 in. D (435 mm × 135 mm × 306 mm).

Weight: 19.8 lbs. (9 kg).

Price: $1,695.

Company Address: 19701 South Vermont Ave., Torrance, Cal. 90502.

(Originally published August 1986)

The CR-7A cassette deck introduces Nakamichi's latest automatic calibration system and offers other Nakamichi firsts as well. The microprocessor-controlled auto-calibration process includes the expected record-sensitivity and bias adjustments, but adds an important element to achieve superior results: The playback-head azimuth is first automatically aligned to the record head to eliminate misalignment as a source of drooping high-end response. Then, the bias adjustments can correct for true response deviations. In conjunction with accurate sensitivity adjustments, the best possible Dolby NR tracking is secured.

The azimuth correction is based upon the detected phase (time) difference between the left and right playback channels with a 400-Hz test tone. Time differences between tracks are the same whatever the frequency, and with squaring circuits, the interchannel time error (ICTE) is easily measured. The ICTE is, of course, directly related to the misalignment, and the system's servo drives to reduce the error to zero (in steps of about a minute of arc). I liked the approach and looked forward to seeing how it would fare in the tests. Nakamichi states that their response corrections have a criterion of ±0.3 dB, and that sounded very good to me, even considering the numerous checks and rechecks of sensitivity and bias during the calibration process. At the conclusion of the procedure, bias and sensitivity data are automatically stored in the CR-7A's memory for that particular tape type, the test oscillator is turned off, user-preferred settings are restored, and the deck rewinds to "0000" and enters record/pause mode, ready to record.

The CR-7A uses Nakamichi's asymmetrical, dual-capstan, diffused-resonance, direct-drive transport for lower audibility of flutter and greater clarity of sound. To gain "smoother tape travel" and "more transparent sound," the capstan drive shafts have a special matte finish and the head assembly includes a pressure-pad lifter. Nakamichi was one of the first manufacturers to use a motor-driven cam to control a number of transport functions, and the three switch cams of the CR-7A inform the microprocessor of system status and tell it how to respond to operator instructions. Automatic slack take-up helps to minimize the chance of damage to tapes.

Quality electronics include direct-coupled recording, line, and headphone amplifiers; a fully discrete playback amplifier with direct coupling from the head; independent power supplies to each circuit, and matched Dolby NR ICs to keep tracking error within ±0.25 dB.

The CR-7A is the first Nakamichi deck to include a real-time counter, which I'm really glad to see—I had almost given up waiting. I think that *every* deck directed at the serious recordist should include at least one real-time counter mode. The CR-7A offers the desirable nicety of both elapsed- and remaining-time display modes. The counter is not a true clock, because it calculates tape time from tape motion rather than measuring time directly. As a result, however, it has the more important attribute of staying calibrated even during fast winding. Time calibration takes about 8 S, which is quite speedy, and recalibration will take place if needed after a fast wind. Calibration is lost if a cassette is removed, but since recalibration is fast, this is of little import. The remaining-time calibration is purposely set

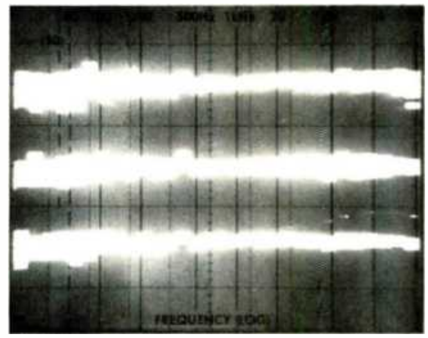


Fig. 1—Record/playback responses for "PN/Music" signal, recorded at −20 dB with Dolby C NR, showing overlaid results from 14 Type I tapes (top), 13 Type II tapes (middle), and 13 Type IV tapes (bottom). See text. (Vertical scale: 5 dB/div.)

so that it reaches "00:00" 5 to 30 S before getting to the actual end of the tape. This helps to avoid the very end of the tape, where faint tape wrinkles caused by the hub clamps reduce recording quality. Also, the zero reading is a reference point for "Auto Fade." With this switched on, a recording will be automatically faded out at the tape end ("00:00") regardless of the actual counter mode. This is a handy feature, especially for those who can't stand the abruptness of a tape run-out.

Control Layout

A look at the front panel reveals other features of interest. The "Power" button is flush with the panel at the upper left; it would be difficult to turn off inadvertently, and that's good. The eject button, some distance below, initiates a smooth opening of the cassette-compartment door. The "Timer" ("Play/Off/Rec") slide switch is below the eject button and above the headphone jack. The compartment door is just to the right of these controls. With the door removed, access to the head and drive assemblies is excellent. Some cleaning tasks are aided by the fact that the unit can be put in play mode without a tape in place, but caution is needed.

Dominating the top middle and right of the front panel is the multi-function display. At the left is the four-digit, three-mode counter display that indicates tape motion, elapsed time, and remaining time. For automatic time calibration, the correct tape length (C-46, C-60, or C-90) must be selected. Little "M" and "S" annunciators under the counter's figures remind the user of the minute/second nature of the readout; there are annunciators for tape length as well, showing the choice that has been made.

To the right of the counter are horizontal, two-channel, peak-responding bar-graph meters, each with 24 segments. All of the segments and some of the scale markings are light tan; "0" up to "+10" are red. The large number of segments and the 5-inch length of the bars make for easy interpretation of levels. Below the meter scales is "Auto

The CR-7A is the first Nakamichi deck to offer a real-time counter, for both time elapsed and time remaining, and I'm very glad to see it.

Calibration," which is always illuminated. When that process takes place, "Azimuth," "Level," "Bias," and "Ready" illuminate in order, just below, showing the status of the calibration procedure. "Ready" stays illuminated at the end of the process, and remains so unless the cassette is ejected or the deck's power is turned off.

Next to these indicators, below the middle part of the meters, is "Tape" with "EX," "SX," or "ZX" illuminated to show the tape type (I, II, or IV) selected manually or in automatic calibration. To the right of these is the EQ readout, indicating 120 or 70 µS. Normally, EQ would be selected automatically along with the tape type, but the CR-7A allows one to switch EQ for particular high-frequency recording needs: 120 µS for more headroom, 70 µS for lower noise.

The NR system choices are shown with "B" or "C" indicators, as well as "MPX Filter." "Subsonic Filter" lights up to show if that is being used. Further to the right are the "Source" and "Tape" annunciators, turned on in accordance with the monitor choice made.

Below the display panel, at the left, are the "Counter Reset" button (which does not affect time modes), the three-position memory switch ("Memory Stop/Off/Auto Repeat"), and the "Counter Mode" button. "Memory Stop" obtains a stop at "0000" with a fast wind in either direction. (Holding in the wind button will get a wind through zero, a desirable configuration.) "Auto Repeat" will get a repeated playing of the entire side of a tape.

Below are nine angled transport-control buttons, each with its own status light, arrayed in three rows. The top three buttons, from left to right, control rewind, play, and fast forward; all have light-green indicators. The second three control pause, stop, and record. The first two have light-green indicators, and record has a red one. The bottom row consists of "Fader" (with a down-pointing arrow), "Rec Mute," and "Fader" (with an up-pointing arrow). All of these have red indicators.

When the record button is pushed, the deck goes into record/pause mode ("Rec Mute" also lights up), and a push of the play button initiates recording. A push of the down fader reduces the record level to zero, and "Rec Mute" turns on again when this is complete. Pushing the up fader returns the record level to where it was. During fading, the intensities of the two fader indicators show the status of the fading. Holding in a fader button gets a faster fade than a single tap. "Rec Mute" mutes the signal while held in, but it does not get an automatic stop, as is obtained on some decks.

Below the middle and right side of the display are small buttons for selecting "EX(I)," "SX(II)," and "ZX(IV)" tape types, as well as "EQ," "Dolby NR," and "Peak Hold." As mentioned earlier, the tape-type and EQ switches are used only when manual choices are desired. The "Peak Hold" circuit gives a 2-S display of peaks at any signal level—even very low levels. This low-level capability is more important than it might seem, for it helps the user to judge all levels similarly. When it is on, "Hold" illuminates just below "Peak" at the left end of the meters.

Below the manual tape switches are the "Tape Length" selector button and the manual "Playback Azimuth" control.

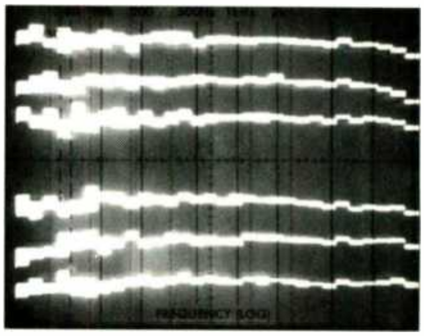


Fig. 2—Record/playback responses to high-level signals with Dolby C NR. Top three traces show response for wide-band pink-noise test signal at "+10" on the CR-7A's meter for Nakamichi EXII, SX and ZX tapes, respectively. Bottom three traces are for record/play of "PN/Music" signal at "+8" on the meter for the same three tapes. See text. (Vertical scale: 5 dB/div.)

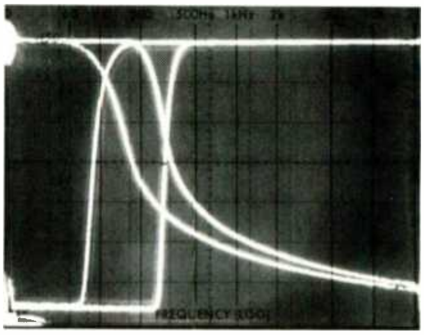


Fig. 3—Action of automatic fader circuits, showing fade-outs (traces descending from left to right) and fade-ins (traces rising from left to right). Left-hand trace of each pair shows action in slow mode; fast-mode fades are to the right. See text. (Scales: Vertical, 10 dB/div.; horizontal, 2 S/div.)

I was struck at once by the outstanding flatness of the record/playback responses, particularly with Dolby C noise reduction.

When recording on the CR-7A, auto-calibration would be the normal route, and azimuth calibration would be part of that process. In playing back a tape recorded on another deck or a prerecorded tape, the front-panel azimuth control allows adjusting the playback head's azimuth to match the actual flux on the tape, setting it for maximum high-frequency output. This control is duplicated on the supplied infrared remote control so that the head alignment can be peaked from the listening position. As soon as the control (front-panel or remote) is turned or pushed to get a change, the meter scales blank out and the topmost meter becomes an azimuth-position indicator with a center arrow. Each step made in adjustment is about 2.5 minutes of arc—acceptably accurate in theory and a great convenience in practice.

To the right are the "Auto Calibration" push bar and its "Reset" button, for use in case of error, and the "Monitor" selector bar for source or tape. Along the bottom right of the front panel are four on/off buttons for "Manual Tape/EQ," "Auto Fade," "Subsonic Filter," and "MPX Filter." The "Tape/EQ" switch has an adjacent red indicator; all of the others have annunciators in the display area.

Along the right-hand end of the front panel, from top to bottom, are level-control knobs: "Master," "Left," "Right," and "Output." The "Master" knob is of medium size, and the other three are small. None have knurling, and the friction is slightly too high for easy turning of the small knobs.

The stereo in/out line jacks on the rear panel are gold plated, which is a nice touch. Two "System Remote' jacks—a DIN socket for transport functions and a mini-jack for the azimuth control—can be used to tie this deck into Nakamichi's CA-7 control amplifier.

I removed the top and side cover to examine the internal construction. After some hours of use, the shielded power transformer was noticeably warm to the touch, but still not hot. There are three soldered-in fuses on the power-supply p.c. board, which also contains the bias oscillator. The large logic p.c. board covers almost two-thirds of the chassis area. Below it is a medium-size p.c. board, and at the bottom is another large card which is close to half-chassis size. The soldering is excellent, but there was some flux noted at hand-wired spots. The logic card is well supported, in general, but there was some springiness noted. The overall chassis construction is rugged and very rigid, with large center and side rails from front to back. The transport was quiet in operation, especially in play.

Measurements

Playback responses with TDK and BASF test tapes were within 1 dB at most points, but there was a greater rise (1.6 to 3.0 dB) at the four highest frequencies of the 70-μS tape. A number of premium decks have shown a rise in this region, although this deck's rise is about 1 dB greater than most others. This comes from Nakamichi's use of playback heads which correspond more closely to the ideal defined in the IEC Standards than to the IEC calibration heads most tape manufacturers use.

Record/playback responses of the CR-7A were checked using "PN/Music" (pink noise rolled off 6 dB/octave at 2 kHz) and a ⅓-octave RTA. I was struck immediately by the outstanding flatness of the responses, particularly with Dolby C NR. Figure 1 shows the -20 dB responses for Type I, II, and IV tapes (top to bottom). Each trace is actually the stored collection of responses for many tapes having a wide range of performance. The Type I tapes included BASF LH-MI, Denon DX1, Fuji GT-I, Konica GM-I, Magnex Studio 1, Maxell XLI-S, Memorex dB, Nakamichi EXII, PDMagnetics FERRO, Scotch XSI, Sony HF-S, TDK D and AD-X, and Yamaha NR—a total of 14 widely different formulations. The Type II tapes included BASF CR-MII, Denon HD6 and HD8, Loran High Bias, Maxell UDS-II and XLII-S, Memorex CDXII, Nakamichi SX, Realistic Supertape Hi-Bias, Sony UCX, TDK SA and HX-S, and Yamaha CR-X—a total of 13 "noncompatible" tapes. The Type IV tapes were BASF Metal IV, Denon DXM, Fuji FR Metal, JVC ME-P, Konica Metal, Maxell MX, Nakamichi ZX, PDMagnetics 1100 Metal HG, Scotch XSMIV, Sony Metal-ES, TDK MA and MA-R, and Yamaha MR—a total of 13 tapes that are not as similar as some have been led to believe.

Table I—Record/playback responses (-3 dB limits).

	With Dolby C NR				Without NR			
	Dolby Lvl		-20 dB		Dolby Lvl		-20 dB	
Tape	Hz	kHz	Hz	kHz	Hz	kHz	Hz	kHz
Nakamichi EXII	10.6	20.0	10.6	21.8	10.6	13.0	10.6	22.3
Nakamichi SX	10.7	13.0	10.7	21.1	10.7	10.2	10.7	22.0
Nakamichi ZX	10.6	21.0	10.6	22.0	10.6	15.0	10.6	22.4

Table II—Miscellaneous record/playback characteristics.

Erasure At 100 Hz	Sep. At 1 kHz	Crosstalk At 1 kHz	10-kHz A-B Phase Error	10-kHz A-B Phase Jitter	MPX Filter At 19.00 kHz
67 dB	60 dB	-100 dB	10	15	-31.9 dB

Table III—400-Hz HDL_3 (%) vs. output level (0 dB = 200 nWb/m).

		Output Level						
Tape	NR	-10	-8	-4	0	$+4$	$+8$	HDL_3 = 3%
Nakamichi EXII	Dolby C	0.12	0.17	0.28	0.47	1.3		$+6.0$ dB
Nakamichi SX	Dolby C	0.07	0.12	0.26	0.76	2.2		$+5.1$ dB
Nakamichi ZX	Dolby C	0.04	0.07	0.13	0.33	0.81	2.0	$+9.2$ dB

I find the results truly marvelous for flatness and consistency, and outstanding for record-sensitivity matching. The vertical spreading of the traces includes statistical effects of the pink noise, any differences in Dolby record-level calibration, any response deviations, and any Dolby C NR mistracking. All of the 13 to 14 responses for each tape type were completely acceptable, but the Nakamichi tapes supplied with the CR-7A (EXII, SX, and ZX) were used for the tests that followed.

I checked the record/playback responses with PN/Music at an rms level equivalent to Dolby level ("+8" meter). They looked so flat (Fig. 2, bottom three traces) that I next fed in, at maximum meter level ("+10"), pink noise that was *not*

The peak-responding meters are just that, except the decay time was short, making "Peak Hold" essential for good metering.

Table IV—HDL_3 (%) vs. frequency using Dolby C NR.

Tape	Level	Frequency (Hz) 50	100	400	1k	2k	4k	6k
Nakamichi ZX	−10	0.06	0.13	0.04	0.05	0.05	0.06	0.08
	0	0.36	0.47	0.40	0.42	0.40	1.0	1.8

Table V—Signal/noise ratios with IEC A and CCIR/ARM weightings.

	IEC A Wtd. (dBA)				CCIR/ARM (dB)			
	W/Dolby C NR		Without NR		W/Dolby C NR		Without NR	
Tape	@ DL	HD=3%	@ DL	HD=3%	@ DL	HD=3%	@ DL	HD=3%
Nakamichi EXII	68.2	74.2	51.3	57.3	68.2	74.2	48.7	54.7
Nakamichi SX	71.3	76.4	54.8	59.9	71.9	77.0	52.7	57.8
Nakamichi ZX	70.3	79.5	53.6	62.8	71.2	80.4	51.5	60.7

Table VI—Input and output characteristics at 1 kHz.

Input	Level		Imp., Kilohms	Output	Level		Imp., Ohms	Clip (Re: Meter 0)
	Sens.	Overload			Open Ckt.	Loaded		
Line	43 mV	>31 V	38	Line	923 mV	770 mV	2.4k	+17.3 dB
				Hdphn.	842 mV	618 mV	18	

rolled off. The results, the top three traces of Fig. 2, show how little roll-off there is even at this very high level. This characteristic is reflected in the excellent figures contained in Table I, showing the −3 dB limits with a sine-wave test tone. The low-frequency response is well extended and very consistent, at both levels and for all three tapes.

Dolby play level indication was high, about 1 dB above the meter-zero level. A number of checks were run to see how well auto calibration aligned the playback head to the recorded flux. Using a 10-kHz test tone for the recheck, there was a consistent 10° phase error between tracks with one tape, which translates to a misalignment error of only 0.3 minute of arc. I used the manual control to try to zero the error with the 10-kHz tone and found that the steps were about 50° of phase in this mode. Nonetheless, I got to within 5° of phase, about 0.15 minute of arc—excellent alignment. The total auto-calibration time was always 15 S or less, with azimuth alignment followed by multiple checking and rechecking of 400-Hz level (for Dolby calibration) versus 15-kHz level (for bias and response). The only time I got an error (indicated by a flashing readout) was when I mistakenly tried to calibrate the Type I tape with manual inputs for Type IV and "70 μS."

The subsonic filter response was 3 dB down at 20 Hz, 20 dB down at 11 Hz, and 30 dB down at 9.8 Hz. The response came back up below this point but was 13 dB down at 7 Hz. The bias in the output during recording was very low. Table II lists a number of other record/playback characteristics. Worthy of note are the excellent 67-dB erasure at 100 Hz and the high separation and crosstalk figures—to say nothing of the low phase error and jitter after auto calibration.

The third-harmonic distortion figures were excellent for all three tapes, and, as Table III shows, those for ZX tape were outstanding. The scan with the spectrum analyzer also showed that distortion was primarily HDL_3, with little evidence of other harmonics. The low level of the distortion made it difficult to measure HDL_3 across the band, and Table IV lists the superior results. Even at Dolby level, distortion was well controlled up to 4 kHz, where tape-saturation effects caused a sharp increase in nonlinearity. Table V provides evidence of how the high maximum output levels of Table III lead to outstanding signal/noise ratios.

Miscellaneous input/output characteristics are shown in Table VI. The line input impedance given is actually a minimum, obtained with all input pots at maximum rotation. With the three pots at a more normal setting, the measured impedance was 83 kilohms—good for minimum loading of other equipment. On the other hand, the line output impedance of 2.4 kilohms is on the high side, particularly if the load is 10 kilohms. A 20-kilohm load would not be a problem. The headphone output drove all phones I tried to very high levels; the output attenuator was needed.

The two sections of the master input-level pot tracked each other within a dB for 60 dB of attenuation, which is excellent. The action of the automatic fader was checked with a 1-kHz tone (Fig. 3) for the two fading speeds, for both fade-in and fade-out. The slow fades are to the left in Fig. 3, and the fast fades are to the right. Although a big contrast exists between the speed of the down-fades and of the much faster up-fades, there is some logic to this approach: The unit fades in fast to be fully up when the music starts, and fades out slowly so the music or applause will trail away to silence. The two sections of the output-level pot tracked within a dB for 40 dB, fairly good. Output polarity was the same as the input in both source and tape modes.

The peak-responding meters met the standards for such meters, with the exception that the 0.7-S decay time was too short. The use of "Peak Hold" appeared essential for good metering. I was not able to verify the accuracy of all the meter-segment thresholds, because they are not tied to specific level figures. Still, the spacing and the results obtained would indicate good dynamic metering. The meter responses were 3 dB down at 10.6 Hz and 20.2 kHz.

There was substantially no measurable change in tape play speed over a range of line power from 110 to 130 V. Over short periods of time, speed variations were on the order of ±0.01%. With selected cassettes, I got flutter values of 0.035% wtd. rms and ±0.055% wtd. peak, very close to the specified values. More typically, I got 0.05% wtd. rms and 0.065% wtd. peak. These are good results but not impressive—and they are noticeably higher than specifications. The fast-wind time for a C-60 cassette was 61 S. There was loose-loop take-up with cassette insertion. Changes in modes and run-outs to stop were all about 1 S.

Use and Listening Tests

The CR-7A owner's manual is clearly written and has helpful illustrations, but some additional detail would aid many users. (I should note, however, that Nakamichi also sent a lengthy technical memo to members of the press, in the form of a news release.)

Sonically, the CR-7A outperformed my reference deck, and the CR-7A's best performance was certainly easier to achieve.

All of the controls and switches were completely reliable during testing and listening. As mentioned earlier, the only problem with auto calibration was a mistake on my part. I really appreciated the wide use of annunciators to show switch status; I had been frustrated so many times in the past with Nakamichi's small, black pushbutton switches—were they in or were they out?

The record, pause, and stop functions all produced light clicks that were down into tape noise with Dolby C NR. I somehow felt personally rewarded with the inclusion of the counter time modes; Nakamichi must have listened to those of us who had pleaded for them. The remote control worked reliably up to at least 20 feet. I put in some prerecorded tapes to try adjusting playback azimuth from my favored listening position, and though about half the tapes were best with the nominal zero setting, others offered a definite opportunity for improvement. Results with the latter demonstrated the value of the Nakamichi approach: There is no other way to match the correction gained by accurate playhead alignment.

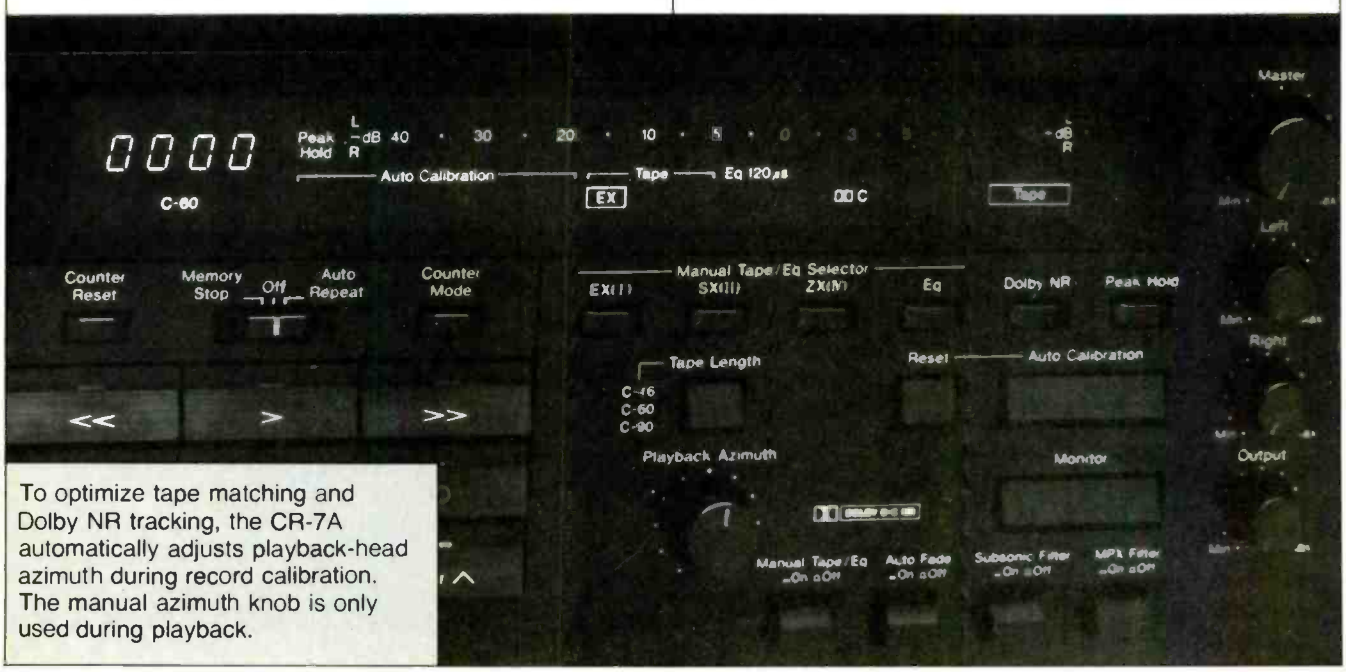
I have mulled over the question of whether adjusting the playback head, as is done in the CR-7A, is essential to get proper alignment with the flux recorded on the tape. Any deck's heads are aligned at the factory, of course—the playback head is adjusted to match a good alignment tape, and the record-head adjustment is made with a no-skew blank tape. The ability to re-adjust the playback head, however, ensures the best possible playback of any tape, from any machine, with whatever skew; it must also be recognized that record-head adjustments can do nothing about correct playback of recordings made on decks that suffer from azimuth errors. I conclude that this feature is very worthwhile, one which I would like to see on more decks.

During recording of various sources, I confirmed my earlier conclusion that "Peak Hold" was essential for the best level metering. I made certain that the peak level went no higher than just below the 3% limits measured during the bench tests. Sources included a number of favorite albums, such as Respighi's *Feste Romane* with Lorin Maazel and the Cleveland Orchestra (Mobile Fidelity MFSL 1-507) and *Buddy Spicher and Friends: Yesterday and Today* (Direct Disk DD102). I did find that with the CR-7A's excellent low-end response, use of the subsonic filter was required with some of the records.

It took me a very short time to decide that the match between the CR-7A's responses with and without Dolby C NR was definitely the best that I have ever heard; I felt similarly about the source/tape comparisons. The frequency response and level matchings accomplished by the auto-calibration system left me nothing to point to as "too much" or "too little." I was very impressed with the CR-7A's ability to retain all of the low bass contained in some of the source material—even at the highest levels. I had found in tests that the flutter was above the stringent specification, but I did not hear any detrimental effects that I could attribute to this. In comparisons with my reference deck, a Nakamichi 582, I judged the CR-7A's sound to be slightly better and its best level of performance certainly a lot easier to achieve.

Overall, the auto-calibration system worked very well indeed and achieved impressive sonic results. The record/playback responses were the best I have measured with Dolby C NR, and in/out and source/tape matchings were outstanding. I wish that the deck had punch-in recording and that the output impedance was lower for some uses, but I'm glad that this unit has counter time modes, manual tape selections, subsonic filter, and manual playback-azimuth control. The price is high, but the Nakamichi CR-7A provides a superlative combination of wide, flat response, low noise and distortion, and a superior auto-calibration system.

Howard A. Roberson



To optimize tape matching and Dolby NR tracking, the CR-7A automatically adjusts playback-head azimuth during record calibration. The manual azimuth knob is only used during playback.

18

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 (effects) quantization.

Processing Programs: 35 preset and 20 user-set.

Harmonic Distortion: 0.002% on main outputs and 0.005% on effects outputs with analog input; 0.003% on main outputs and 0.005% on effects outputs with digital input.

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

S/N Ratio: 110 dBA for main and 94 dBA for effects with analog input; 110 dBA for main and 105 dBA for effects 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: 6722 Orangethorpe Ave., Buena Park, Cal. 90620.

(Originally published November 1988)

When Yamaha introduced the DSP-1 digital sound field processor, I was among many who 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. Four new movie-theater modes 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. A program menu is used 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 inputs and a direct digital input. The digital input allows direct effects processing of CD or DAT signals and eliminates one stage of A/D 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 effects 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 from the effects 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.

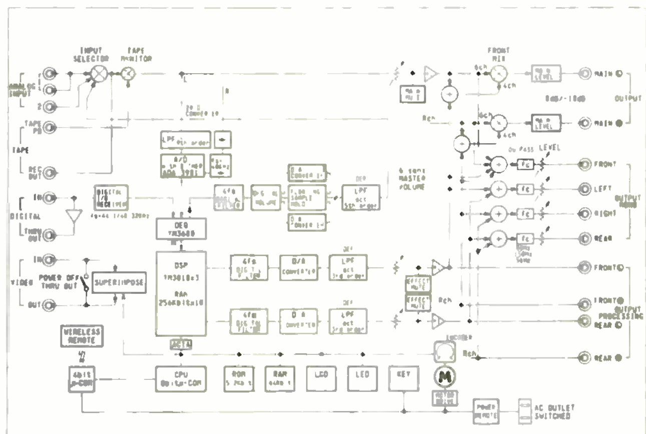


Fig. 1—Block diagram. Note that all surround processing takes place in the digital domain.

Control Layout

The on/off "Power" pushbutton is at the lower left of the front panel. Above it is a display area that extends from the panel's 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 muting 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. Bright yellow LEDs are 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 analog input 2 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 selectors are the "Tape Monitor" switch and the down/up "Program" rocker. A red LED illuminates when the monitor switch is on. 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 preset "20" 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 are stereo (L/R) pairs for "Analog Input" (two sets for "1" and one set for "2"), "Tape" inputs and outputs, and the "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 Processing" jacks selects "4 ch" or "6 ch" to match the system configuration. The normal system has four separate effects channels in addition to the two main channels. When the system will have just two effects 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 Hz, 150 Hz, and 5 kHz. The pot knobs are very small, but knurling makes them easy to turn. As Fig. 1 shows, appropriate outputs for the effects channels 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 effects channels.

To the right are the "Digital" "In" and "Thru/Out" jacks. This configuration allows sending the digital signal from a CD player or DAT recorder 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 neavy top cover to get a look at the internal construction ana 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 but-

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

ton (discussed below) is usec 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 re-creating 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") comprises: "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" change 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 it enables system balancing in combination with "Preset" and the built-in pink noise source. (The "Measurements" 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 any change from 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 muting 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" display showed "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 causing the display's "Full" legend to turn on. The clipping point, however, was greater than 10 dB above where "16 bits" appeared with the same tone 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 effects 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 control 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 the 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 volume controls tracked each other within 1 dB over the whole range, which is excellent. The two rear-channel volume controls tracked within 1 dB for about 50 dB of attenuation, which is quite good. The effects-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.

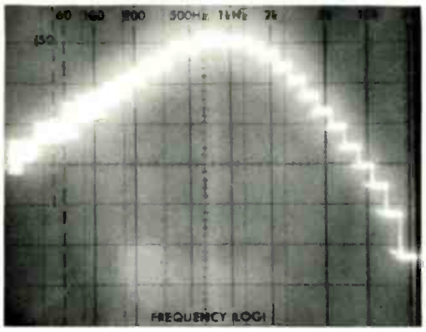


Fig. 2—Balance test signal generated by the DSP-3000. Though called "pink noise," its frequency content is actually optimized for speaker balancing. See text.

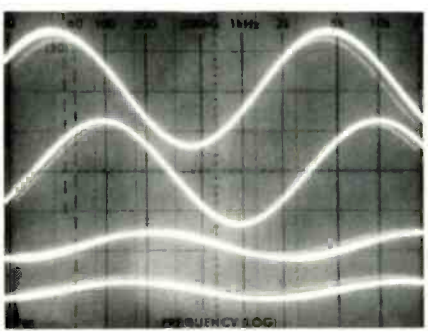


Fig. 3—Output from effects channels in "Movie Theater/Adventure" mode, with 742.6-Hz test tone. Traces are (top to bottom): LF, RF, LR, and RR. Changing the test frequency would change the phase and amplitude relationships between the channels. See text. Vertical scale: 2 V/div.

The effects 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. Checking 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 effects 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 effects 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 outputs of the four effects channels 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). The initial delay can be set in 1-mS steps from 1 to 150 mS, while the setting for reverberation time has a range from 0.3 to 10.0 S, with 0.1-S steps. The level of reverberation 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 can 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 3 to 6 feet from the front wall, with the front effects 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 Design subwoofer, and a Yamaha FM tuner, videodisc 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.

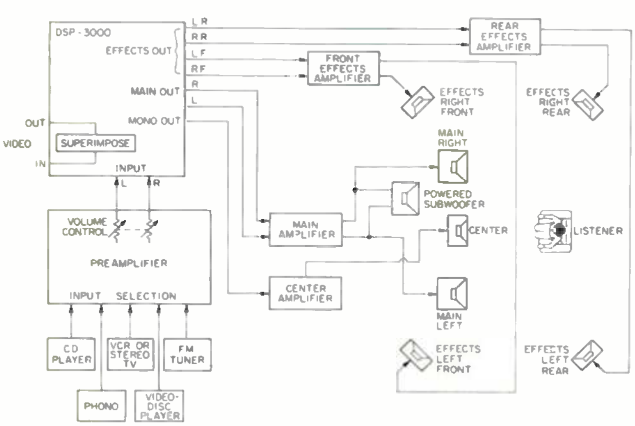


Fig. 4—Layout of the sound system used in evaluating the DSP-3000.

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 levels exactly to the point which would yield full 16-bit processing.

Using the built-in noise source to match main and effects levels, I found that I had to switch the main output to −10 dB to have the desired level range. Throughout my listening tests, I shifted effects 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.

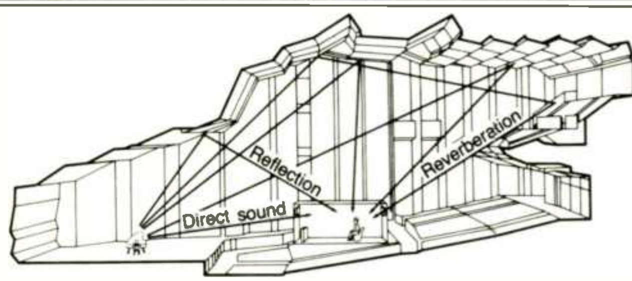
In the listening evaluation, I purposely picked sources to match the various programs, and then tried other programs if that seemed worthwhile. Unless stated otherwise, CDs were the sources.

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.

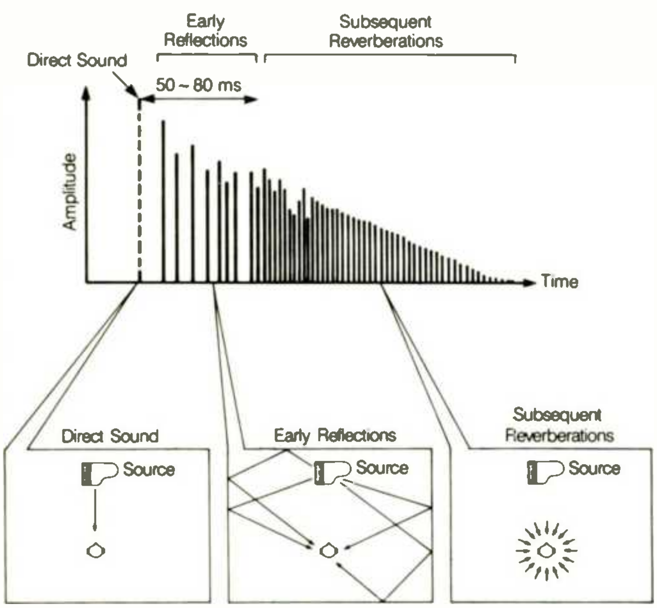
With Dvořák's Symphony No. 9 with Solti and the Chicago Symphony (London 410116-2), I preferred Hall G in the U.S.A., but I also liked Halls B and C in Europe and Live Concert A (program 5). Some overtures by Elgar, with Gibson and the Scottish National Orchestra (Chandos CHAN-8309), sounded best with 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" choice. 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 playoff 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.

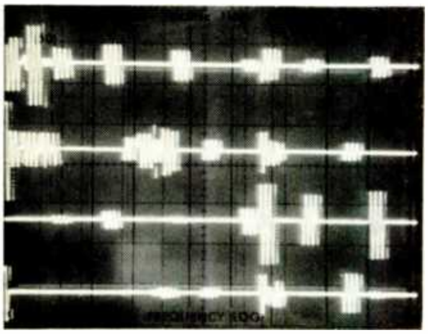


Fig. 5—Output from effects channels in user-modified "Presence A" program, with three-cycle, 700-Hz tone burst. Traces are (top to bottom): LF, RF, LR, and RR. See text. Scales: Vertical, 1 V/div.; horizontal, 11 mS/div.

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*

19

FOSGATE DSM-3610 PRO-PLUS SURROUND PROCESSOR

Manufacturer's Specifications

Static Separation: Better than 35 dB side to side, center to surround, and surround to front. Typically better than 50 dB from center to surround and surround to center.

Dynamic Separation: Sufficient for instantaneous localization in all directions simultaneously.

Main-Channel Distortion: 0.05% at 2 V output.

Frequency Response: 5 Hz to 35 kHz, ±1 dB.

S/N Ratio: 90 dBA, re: 1.5 V.

Surround-Channel Distortion: 0.3% or less.

Dolby Surround Frequency Response: To Dolby Laboratories specifications.

Surround-Channel S/N: 85 dBA, re: 1 V.

Subwoofer Frequency Response: 5 to 80 Hz, with roll-off at 12 dB/octave above 80 Hz.

Bass EQ: Up to 18 dB boost.

Input: 100 mV to 3.5 V, 75 kilohms.

Output: Up to 4 V, 1.5 kilohms nominal.

Dimensions: 17¼ in. W × 2¾ in. H × 11 in. D (43.8 cm × 7 cm × 27.9 cm).

Weight: 9.8 lbs. (4.5 kg).

Price: $1,429.

Company Address: P.O. Box 70, Heber City, Utah 84032.

(Originally published March 1989)

Jim Fosgate and Peter Scheiber have been involved for many years in creating designs and products for various forms of surround sound. The result of their latest collaboration, the Fosgate 360° Digital Space Matrix DSM-3610 Pro-Plus, is an advanced separation-enhancement system.

Sophisticated digital control technology allows the time constants of the logic steering circuitry to change constantly with the dynamics of the source. This is true whether the material is encoded with Dolby Surround or is regular stereo. The attack and release times of the logic-control signals are automatically adjusted in response to complex material, thereby preventing IM distortion, pumping, or breathing effects. These times can be very short when called for.

An analog time delay is used, which the makers feel has a more natural sound than digital delays. The Pro-Plus system includes a modified Dolby B NR circuit, to encode 10 dB of noise reduction in addition to Dolby Surround's standard 5 dB. Fosgate states that the combination "results in a time-delay system with the quietness of digital and the natural sound of analogue."

The DSM-3610 offers four operating modes: "Mono," for synthesized stereo surround from monaural sources, plus "Regular," "Medium," and "Wide" surround modes, all of which are compatible with Dolby Surround. It has input switching for four audio/video sources, plus A/V tape-monitor connections. Controls for input level and balance, surround level and delay, and bass EQ (adjustable from 0 to +18 dB) are on the front panel. The supplied infrared remote control can change overall volume and main/surround balance, mute system output when needed, and restore all factory-set adjustments with the touch of a button. (Adjustments for more exact level matching to external amplifiers, should that be necessary, are available inside the DSM-3610.)

The unit has outputs for main stereo, center front, and subwoofer channels as well as for the left and right side and left and right rear surround channels. The surround delay is continuously adjustable from 15 to 30 mS.

Control Layout

Along the left side of the front panel are 11 pushbuttons, each with a large LED indicator, in groups of five, two, and four. These buttons require a firm push to ensure latching. Light-touch switches may be in vogue these days, but I have seen such switches fail with time and not work no matter how much pressure was applied. The switches used by Fosgate have contacts that wipe across each other in operation, which promotes long-term reliability.

The first group of five pushbuttons is for "AV Source." The buttons labelled "One" through "Four" have green LEDs and are mechanically interlocked. "Tape Monitor," the last of the five, is not interlocked with the others and should not be. When it is on, its yellow LED cautions the user that the DSM-3610 is in monitor mode. When activated, all of these "AV Source" buttons switch both video and stereo audio.

The next two buttons to the right are "Logic" (red LED) and "Center Ch" (yellow LED). "Logic" engages the Fosgate Pro-Plus steering logic. "Center Ch" activates the center-front channel to feed a center amplifier and speaker.

The next group of switches is for "Sound Stage Width." These buttons, each with a green indicator, offer choices of "Mono Enh," "Regular," "Medium," and "Wide." "Mono Enh" is used with monaural sources and enhances them by synthesizing a surround effect. "Regular" provides better-than-theater Dolby Surround effects from encoded sources and provides a distant perspective for stereo listening. "Medium" yields a mid-hall perspective with stereo or surround-encoded material. It omits the normal Dolby Surround delay and response-restricting, 7-kHz filter from the side (but not the rear) channels. "Wide" is used to get an up-close, "you are there" perspective from a variety of sources.

In the center of the front panel, just to the right of the pushbuttons, are a number of LEDs and the remote sensor. From left to right are: "Dialog" (red LED), "Surround" (red LED), "IR Sensor," "IR Receiver" (red LED), and "Input Level" (three green, one yellow, and one red LED, side by side). When processing stereo material, the "Dialog" and "Surround" LEDs flash on and off in accordance with the relative center and surround content of the program material. The small, round infrared sensor is inset into the panel to protect it from possible damage. Its LED flashes rapidly whenever the remote control is used, confirming that transmission is being received. The "Input Level" LEDs form a simple, left-to-right level meter. The leftmost green LED is always on when the unit is powered, the yellow LED indicates caution against higher levels, and the red LED calls for level reduction.

Further to the right are five rotary controls: "Input Level," "Input Balance," "Bass EQ," "Surround Level," and "Surround Delay." Below each are guiding labels at the counterclockwise and clockwise ends of rotation. The labels are, respectively, "Min/Max," "Left/Right," "Bypass/+18 dB," "Min/Max" and "15 mS/30 mS." Each of the medium-sized knobs has good knurling and an obvious white index line, both of which are very helpful. It would be even better if each index line extended onto the face of the knob: When a knob is at either extreme position, its index line cannot be seen from above. At the far right is the power on/off switch. The panel's gold legends are hard to see on the black background if the light is somewhat dim.

On the back panel are four groups of gold-plated phono jacks. From right to left, the first six are the "Video Switch" group, labelled "1," "2," "3," "4," "Tape," and "Out." The "Tape" jack allows connecting the video output from a VCR, so its output is looped through when the tape-monitor switch on the front panel is used. The "Out" jack will feed the selected source to a video monitor. The second group consists of "1" through "4" stereo pairs for the "Audio Inputs." Next is the "Tape Recorder" group, which has audio "Tape Out" and "Tape In" stereo pairs. The "To External Power Amplifiers" group has stereo pairs for "LF/RF" (main), "LS/RS" (side) and "LB/RB" (back surround), and single mono outputs for "CF" (center front) and "Sub" (subwoofer). To the right of the unit's power cord are two unswitched a.c. outlets. The fuse-holder below the cord has a flat cap with a screwdriver slot. This good design makes it possible to check a fuse externally but does not make it that easy to diddle. A label calls attention to the fact that this Fosgate unit has been treated with Tweek, to prevent corrosion and to maintain good contacts at connections.

Removing the top cover revealed a chassis-size p.c. board having an open and very neat layout. Some parts numbers are shown, and many components and sections are identified by function. The eight user-adjustable trim pots are very clearly marked, and an accompanying statement warns the user about changing any other controls. I noticed that the other, factory-adjusted trim pots were marked and staked in place by small dabs of red lacquer. Three fairly large black boxes (literally) are the "Pro-Plus D-3 Digital Control Voltage Generator," "Pro-Plus MX High Separation Matrix," and "V-1 Electronic Volume Control." Many parts of the highest quality were in evidence. The soldering on a small vertical board was excellent; I did not remove the bottom cover to look at the soldering on the main board. The power transformer, mounted on the side rail and the board, was fairly hot to the touch after hours of operation. The chassis was good and rigid without the top cover—more so, of course, with it back in place.

The remote control is very simple in comparison with many others, and its functions are easy to understand. This

The DSM-3610 was successful in placing voices with the on-screen characters while maintaining spread in the music and effects.

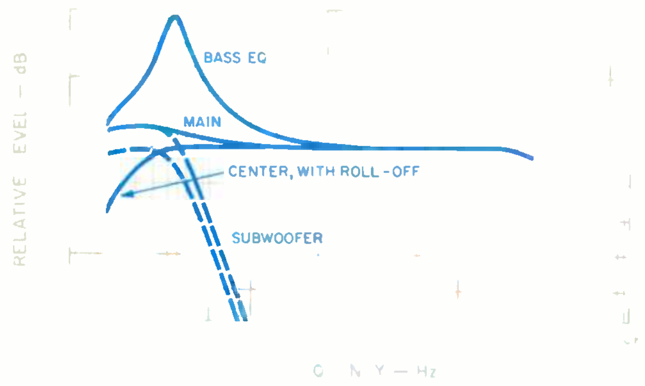


Fig. 1—Swept-frequency response curves for various channels and settings of the DSM-3610; see text.

simplicity could be a considerable advantage for many users, although others will miss being able to switch modes from the listening position. The remote's two "Vol" buttons increase or decrease overall level, while the two "Bal" buttons shift the front/back balance. A push of "Ref" returns volume and balance settings of the DSM-3610's voltage-controlled amplifiers to factory-set references. "Cue" drops the volume to a low level when desired, such as when answering the telephone. A second push, or a touch of either volume button, restores the set volume.

Measurements

I should first note that all measurements were made after I had completed the listening tests that are discussed later.

The main-channel frequency response (Fig. 1) rose slowly as the frequency decreased below 1 kHz, reaching +1 dB at 100 Hz and close to +3 dB from 40 down to 20 Hz. It then rolled off to reach 0 dB at 4.3 Hz and −3 dB at 2.7 Hz. Above 1 kHz, response was flat to nearly 20 kHz, then rolled off to −0.8 dB at 20 kHz. The −3 dB point was reached at 45.9 kHz.

The center channel's response was basically the same as the main channel's, including the low-end boost. Figure 1 also shows the low-frequency response of the center channel with its internal roll-off switch on. This roll-off would be recommended for center speakers having poor bass capability or for a better overall balance when a subwoofer is used. The subwoofer output rolled off above 50 Hz, reaching a slope of 18 dB/octave at about 80 Hz. The subwoofer internal trim pot had a range of 25 dB. Figure 1 shows the subwoofer channel's response with this pot adjusted to match the main channel's level at 40 Hz (+3 dB) and with the pot adjusted to match the main channel's 1-kHz output (0 dB). The bass EQ's boost peaked at 58.5 Hz, with a maximum rise of 17.2 dB; this is in addition to the main channel's normal response boost of 2 dB or so at that frequency. The side channels were −3 dB at 30 Hz and 7.7 kHz, and −10 dB at 9.7 kHz in "Regular." In the "Wide" operating mode, the −3 dB point moved out slightly, and −10 dB was reached at 12.6 kHz. The back surround-channel responses for all modes were close to the "Regular" side-channel response.

Harmonic distortion for 1 V at 1 kHz was 0.03% in the main channels, falling to 0.028% at 20 Hz and rising to 0.3% at 20 kHz. At 0.5 V, a much more likely voltage, the distortion at 20 kHz was 0.12%, which is much better. The surround channels had 0.05% distortion for 1 V at 1 kHz. The 20-Hz figure was 0.15%, and the high-frequency distortion was 0.3% just before the roll-off point.

With "Ref" volume and balance, the S/N ratio of the main channel was 93.3 dBA referred to 1 V, and this would be close to typical over a range of adjustments. With volume at maximum and balance all the way to the front, the ratio decreased to 80 dBA, which is a worst-case figure. The side channel's S/N was 91.7 dBA with reference volume and balance. The back surround channel's S/N varied from 80 to 90 dBA, depending on particular settings. This ratio was typically 85 dBA with reference volume and balance and with the surround-level pot at 1 o'clock.

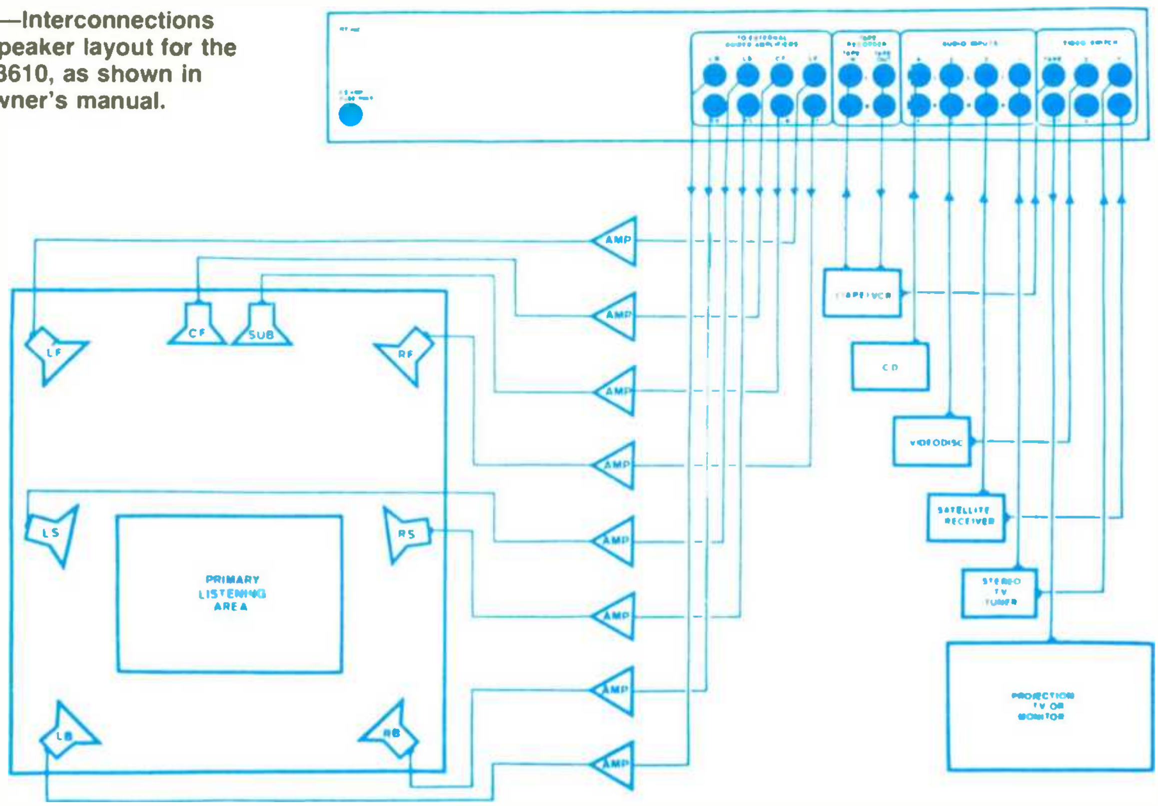
The maximum input level for a 1-kHz test signal was greater than 31 V. With the input-level pot wide open, 0.196 V would just turn on the level meter's red LED; actual waveform distortion appeared 4 dB above that. I fed in a 5-kHz tone burst to check the response of the LED meter and set the continuous level 1 dB above where the red LED turned on. I was quite impressed to see that the LED was still flashing brightly with a burst as short as 10 mS. In fact, it was still flashing, albeit faintly, with bursts as short as 0.4 mS. The decay time was about 250 mS, somewhat faster than a VU meter. The DSM-3610's little meter may not look like much, but it is an important and well-implemented feature.

Maximum output, defined by the onset of clipping, was 6.7 V with the internal level-adjustment trim pot turned up. This voltage is much higher than is called for by the sensitivity of any power amplifiers I know of. There was, therefore, no need to keep the trim this high, so I returned it to the factory setting. The input impedance was a satisfactory 22 kilohms with the input pot at maximum and a good 39 kilohms with it at midpoint, a more likely position. The output impedance was 675 ohms, which is a very good figure. The surround delay time could easily be set anywhere from 10.8 to 30.0 mS.

Using a monaural source, I adjusted "Input Balance" to get a minimum level in the surround channels. With this setting, the left main-channel output was just 0.3 dB higher than the right. The left and right sections of the input-level pot tracked within 1 dB from 0 to 40 dB of attenuation, which is fairly good. Remote-control volume and balance tracked within 1 dB for about 25 dB over their total 40-dB range. From maximum volume and balance all the way front, "Ref" reduced the main channel's volume by 18 dB, including a

I was pleasantly surprised at how well the combination of music and announcements sounded. Some processors can't manage both at once.

Fig. 2—Interconnections and speaker layout for the DSM-3610, as shown in the owner's manual.



14-dB level drop plus a 4-dB balance shift from front to back. "Cue" reduced the maximum volume by 37 dB.

Use and Listening Tests

The DSM-3610 owner's manual starts with a quick hookup guide for those who can't wait to hear something. The final step recommends reading the entire manual, which I feel is a must for the best use of this Fosgate processor. There is good discussion on front-panel controls, rear-panel connections, and the possible sources of audio and video, plus some suggestions on which modes to try. The installation instructions are quite good, but they would be improved if references to the illustrations at the back of the manual were included. It's good to instruct the user on checking loudspeaker connections for consistent polarity relative to amplifier outputs. Missing, however, is the caution that some power amplifiers invert polarity, and some do not. The section on making internal adjustments is well written and includes the steps needed to get personalized reference-level settings for future use of "Ref"; here, though, users should have been referred to the back-of-the-manual illustration of these pots. The sections on "Optimizing System Performance" and "In the Event of Difficulty" are succinct, lucid, and pertinent.

My reference processor for the listening/viewing tests was the Yamaha DSP-1. Other equipment included a Yamaha AVC-50U for input switching and main-channel amplification, a Yamaha TX-900U AM/FM tuner, a Magnavox FD1041BK CD player, an Akai VS-M930U-B VHS Hi-Fi VCR, a Sanyo VCR-7200 Beta VCR, a Yamaha LV-X1 videodisc player, a Soundcraftsmen DC2214 octave-band equalizer, a Yamaha M-35B four-channel amplifier for the side and surround speakers, a Lafayette amp for the center channel, a QSC amp for trying other speaker locations, and speakers from JBL, Dynaco, and Ramsa. The Akai VHS Hi-Fi VCR was used as the MTS stereo-TV decoder. I used a patchbay that facilitated making fairly fast changes between the Fosgate DSM-3610 and the reference Yamaha DSP-1.

Figure 2 is a reproduction of one of the illustrations in the DSM-3610 owner's manual. I did not use the tested unit for switching inputs, but the outputs, amplifiers, speakers, and arrangement in Fig. 2 match what I did.

For my first listening/viewing test, I watched a VHS Hi-Fi version of Paramount's *Planes, Trains and Automobiles*, with Steve Martin and John Candy. Some of the balances weren't quite what they should have been, but I hadn't taken the time to adjust them according to the manual. I did, however, set the surround delay to correspond to the listening area and came to a number of conclusions fairly quickly: The results were best with "Logic" and "Center Ch" on; the sound effects and background music were well integrated into the overall sound; "Medium" was the preferred mode; I couldn't sit close to a back surround speaker, and the character of the dialog was very good.

The DSM-3610's quality and performance should interest those who want high-quality home-theater sound together with enhanced stereo music.

Even though I had more setup adjustments to do, I made a fast comparison with the Yamaha DSP-1 and confirmed my judgment that the DSM-3610 delivered superior dialog from this movie. The improvement in the sound of the dialog led me to re-aim the center speaker so that my preferred listening position would be more on its axis. Then I followed the procedures in the manual to get better level balances among all the speakers. The next source was the NBC movie, *A Stoning in Fulham County*, with Ken Olin, Jill Eikenberry, and Ron Perlman. The results were very good with both "Logic" and the center channel on, and were not as good with either or both off. I was able to set the center channel's level exactly where I wanted for good, centered dialog without losing a good spread in music and effects. Because I had matched levels well, particularly between the side and back speakers, it was much more difficult to localize the rear surround speakers than it had been before. I determined that the remote control was effective up to at least 25 feet and over 30° off axis.

Aliens, with Sigourney Weaver, on Showtime, had good dialog centering even when the center channel was off, but I preferred it on. There were good, pertinent alterations in the sound field with changes in the scene. Poor surround systems can produce changes that are interesting but wrong for what appears on the screen. One scene was particularly exciting. A warning beep was sounding, and I suddenly realized that I was getting tense from the action and from being surrounded by this persistent tone—very effective. A rented VHS tape, *From Beyond*, with Jeffrey Combs and Barbara Crampton, required the monaural setting and "Logic" off. The results were fairly good—better than I expected.

The Warner Home Video *Ladyhawke* videodisc, with Rutger Hauer, Matthew Broderick, and Michelle Pfeiffer, produced the best sound I'd yet heard from this setup, to say nothing about the best picture. "Logic" and the center channel were both on, and "Medium" was the preferred mode—especially for the music, which I really like. There was one short section where there was some soft popping, but it disappeared with the steering logic off. The popping did not occur at any other point, so I suspect the disc itself was responsible. This conclusion was reinforced when I played Paramount Home Video's *Rustlers' Rhapsody* videodisc, with Tom Berenger. The results were similarly excellent and without any popping. I have commented in the past about other systems that spread stage-center voices out in space until they seem disembodied. These two discs helped to emphasize the DSM-3610's success in placing voices *with* the characters, while maintaining spread in the music and effects.

When listening to my favorite FM station, I preferred "Medium" or "Wide" mode, depending on the music. I left "Logic" in and the center channel on most of the time and was pleasantly surprised at how well the combination of spoken announcements and played music sounded. Previously tested surround processors offered the choice of good voice quality or good music sound—not both.

I used CDs for most of the music-source listening. The well-known Pachelbel Canon in D Major, performed by the Jean-Francois Paillard Chamber Orchestra, was best using "Medium," with center and logic off. The sound had good, smooth quality, but, overall, it was not a match for what was possible with the Yamaha DSP-1. I came to similar conclusions for other pieces on this Erato CD, entitled *Pachelbel: Canon/Albinoni:Adagio* (ECD-55018).

For Mozart's Symphony No. 39, played by the Bamberg Symphony Orchestra with Eugen Jochum (Orfeo C045901A), the "Regular" and "Medium" modes were both good. *Music of Wagner* (Minnesota Symphony Orchestra with Neville Marriner, Telarc CD-80083), Schubert's *Death and the Maiden* (Amadeus Quartet, Deutsche Grammophon 410024-2 GH), and some Charpentier motets (Concerto Vocale, Harmonia Mundi HMC-901149) were all best with "Medium" selected. Dire Straits' *Brothers in Arms* (Warner Bros. 25264-2) was especially good with "Wide." The center speaker was very good for pointing up vocals on this and other pop/rock CDs. "The Atlantic Records 40th Anniversary Show" on HBO featured, among others, Phil Collins, Sam Moore, The Bee Gees, The Rascals, and Dan Aykroyd. The center channel was definitely needed for good vocal centering and presence. I thought "Wide" mode was best for both music and a "being there" audience sound.

Although I had wished for more features on the remote control during setup and early testing, I did not feel so limited after some use. I suspect that many audiophiles would have a similar experience: After learning what modes and control and switch settings are best for particular sources, those choices will be made when selecting the source while at the equipment. That's also the time to check input level and change bass EQ, if necessary.

In my own listening, I thought that the sound was good and full with bass EQ at zero. I did not judge the bass to be excessive and was a bit surprised at the response boost revealed in the later measurements. If a turntable is used with this system, a subsonic filter may be needed to reduce possible rumble. There is a slight lag when changing volume or balance with the remote, but the shifts are desirably smooth. "Cue" requires a short hold on the button—a quick tap is not long enough for response, even though the front panel's "IR Receiver" light goes on. I liked the way the muting went on and off because the level changed very quickly but smoothly—not abruptly, as is typical.

I do feel that Fosgate's combination of the 360° Digital Space Matrix and the Pro-Plus steering logic is successful. This is particularly true for movies—whether broadcast, on videocassette, or on videodisc. Music performances on TV, including music videos, also benefited from the performance of the DSM-3610. In comparison to the reference processor, however, the Fosgate was audibly less successful with classical music—although it did provide a better compromise for some broadcast music programs with spoken commentary.

One of the tested unit's strong points is its provision for side speakers, which secure a general improvement in the smoothness of the sound field. The side speakers also enlarge the possible listening area and make the back surround speakers less likely to be localized.

The Fosgate DSM-3610 has a high price, but its quality and performance make this sound processor of interest to those who want really high-quality home-theater sound and better-than-stereo music reproduction. *Howard A. Roberson*

20

SHURE HOME THEATER SOUND HTS 5300 SURROUND DECODER

Manufacturer's Specifications

Frequency Response: Front left, center, and right, 20 Hz to 20 kHz, ±0.5 dB; subwoofer, −3 dB at 80 Hz with 12-dB/octave roll-off; surround, 50 Hz to 7 kHz, −3 dB (per Dolby Surround specifications).

Input Sensitivity: 0.25 V.

Maximum Input and Output Levels: 4.0 V.

Range of Input-Balance Control: ±9 dB.

Range of Output-Level Trim Pot: 20 dB.

Impedance: Input, 75 kilohms; output, 5.5 kilohms.

Distortion: Main channels, 0.1%; surround channels, 0.3%.

S/N Ratio: 90 dBA re: 1 V, with volume controls centered.

Signal Polarity: Noninverting at all outputs.

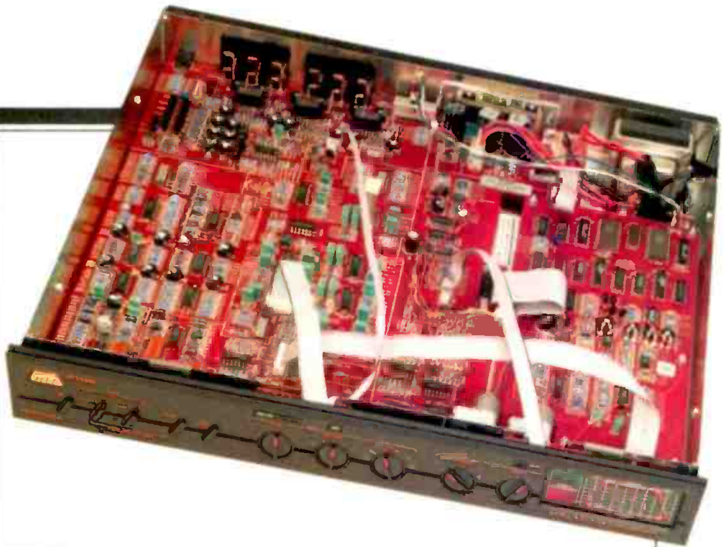
Surround Delay: 16 to 36 mS, in 4-mS steps.

Dimensions: $16\frac{13}{16}$ in. W × $2\frac{3}{8}$ in. H × $15\frac{1}{16}$ in. D (42.7 cm × 6 cm × 38.2 cm).

Price: $1,250.

Company Address: Shure HTS, 222 Hartrey Ave., Evanston, Ill. 60202.

(Originally published July 1989)



The HTS 5300 is the surround-decoder part of the Shure HTS Theater Reference System. The complete $9,600 system contains all that is needed for a surround-sound installation except for sources and miscellaneous accessories. Besides the decoder, the system includes three HTS 50SPA power amplifiers, one HTS 50CF center-front loudspeaker, one HTS 50SW subwoofer, and four HTS 50LRS loudspeakers. The latter four speakers are used for the main left and right stereo channels and the two rear surround channels.

My testing was restricted to the decoder, but it is worthwhile to discuss the entire Theater Reference System. This is truly the result of a system design approach: It is *not* a collection of already available components stuck together just to have all the parts. The configuration, of course, revolves around what the decoder does with the proper sources, but I'll go into more detail on that later. At this point, I'll restrict my comments to stating that the decoder's outputs consist of the normal stereo pair plus one each for center-front and subwoofer and a pair for the surround channels. The three two-channel power amplifiers drive the six speakers.

The typical home surround system has been somewhat of a hodgepodge, with amplifiers and speakers used from previous systems—perhaps with additional purchases made to get all the channels needed. Often, the new amps and speakers are not the same as the original ones, for various reasons. As far as I know, Shure HTS is the only manufacturer which offers a complete system with correlated designs. The discussion that follows will not only detail what it consists of but should also help explain the interrelationships among the components of a surround system.

The HTS 50SPA is a signal-processing power amplifier with switch-selectable operating modes to match the speaker complement; it delivers 100 watts per channel. Each of the two channels has a level control with useful decibel scaling and a six-position "Operational Mode" rotary switch. The knobs are rounded discs with large slots which reject casual diddlers but accept large coins or a strong thumbnail for turning. The first five settings are "Flat," "LRS," "LRS$_x$," "CF," and "CF$_x$." The sixth position is "SW" for channel 1 and "Bridged" for channel 2.

In "Flat," the amp's rated response is ±0.5 dB from 20 Hz to 20 kHz, and there is no processing in the signal path except for a defeatable clipping-protection circuit. The "LRS" setting switches in an 80-Hz low-frequency cutoff for use with the HTS 50LRS (left/right/surround) loudspeakers in a system with a subwoofer. The "LRS$_x$" position, on the other hand, is for the same speakers in a system without a subwoofer. In this case, the response extends down to 60 Hz. The "CF" output has an 80-Hz roll-off to match the response of the HTS 50CF (center-front) loudspeaker in a system with a subwoofer. With "CF$_x$" engaged, the center-speaker response is extended down to 55 Hz for a nonsubwoofer system. The output of channel 1 in the "SW" mode matches the HTS 50SW subwoofer, covering only the frequencies from 33 to 80 Hz and adding a controlled low-frequency boost. The "Bridged" setting of channel 2 reconfigures the amplifier into a single-channel unit delivering 250 watts into 8 ohms. In this mode, the channel 1 selector determines the response of the amplifier, matching it to any of the HTS 50 loudspeakers. The amplifier has circuitry to limit cone excursion, which is particularly important in this mode. Having the ability to instantly configure response to suit specific applications is very appealing to me: Unwanted energy is not fed to any particular speaker, and all of the amp's power is available for the band selected. The amplifier can, of course, be used with any brand of speaker, though preferably with its mode switch in the "Flat" position, which provides only overload protection.

All of the HTS 50 loudspeakers are rated to handle 200 watts peak program material and 100 watts nominal amplifier power. With each of these speakers, the HTS 50SPA amplifier reduces system distortion by controlling cone ex-

Shure HTS seems to be the only manufacturer to offer a complete surround *system* of correlated decoder, amp, and speaker designs.

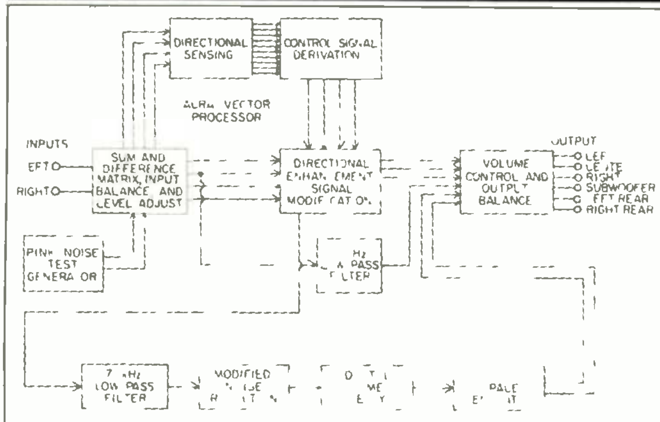


Fig. 1—Block diagram of signal processing in the HTS 5300.

cursion at low frequencies and very high sound levels. All speakers have double magnet systems to reduce interference with TV picture convergence and purity. "In a high-performance home theater sound system, the loudspeaker used in the center-front position is subjected to the most rigorous performance demands, particularly in regard to output capability," says the company.

The HTS 50CF center-front speaker has two 6½-inch low-frequency drivers and a single 1-inch, soft-fabric, damped-dome tweeter. The response of the HTS 50CF runs from 55 Hz to 18 kHz in a system where subwoofers are not used, so the amplifier is set for extended response ("CF_x").

The HTS 50LRS, used for the left and right surround speakers, is generally similar but has only one low-frequency driver—which is sufficient because of the lower power demands of the surround channels. This speaker's response goes from 60 Hz to 18 kHz in systems where the amplifier is set at "LRS_x" because there is no subwoofer. Both the center-channel and surround speakers play only the frequencies above 80 Hz when a subwoofer is used; the amplifier channels are set to "CF" and "LRS," respectively. The HTS 50CF and HTS 50LRS speakers have two-way, fourth-order crossovers with corrections for impedance equalization and midband response. The HTS 50SW subwoofer has a 12-inch transducer mounted in a fourth-order vented box. Combining this with the second-order, high-pass 33-Hz filter of the HTS 50SPA (that is switched in when the amplifier is in "SW" mode) yields sixth-order dynamic tuning. The amplifier also has a second-order, low-pass filter at 80 Hz; the total 80-Hz filtering is fourth-order when the HTS 5300 decoder's subwoofer output is used.

Acra-Vector Decoder Circuitry

Figure 1 is the block diagram of the HTS 5300 decoder's signal processing. This latest Shure HTS Acra-Vector decoder has 80% more sensing points than previous models, for smooth and accurate imaging. Acra-Vector logic emulates the Dolby Stereo theater decoder and uses directional-enhancement (steering-logic) circuits which recover the original four recorded channels with a high degree of separation between adjacent channels. On complex source material, the HTS 5300 is more capable in the "proper enhancement of multiple simultaneous sound sources," says the manufacturer. The HTS 5300 has independent control signals for enhancement of left-right, center-surround, and similar signal oppositions. As a result, according to the company, there can be "simultaneous enhancement of two opposite directions at one time." Low-level directionality is also more accurately detected.

The HTS 5300's digital time delay has twice as much memory as the previous Shure HTS decoder; this has reduced noise up to 9 dB in the surround channels. The decoder uses the Shure HTS Acoustic Space Generator for "clean, spacious theater and concert hall ambience extraction" and a "full spatial field with a minimum of surround speakers." Sonic performance has been improved by extending the headroom at high frequencies. The HTS 5300's front panel has Shure's Image Analyzer display, which shows sound fields in high contrast. A built-in precision test generator steps automatically from channel to channel for easy balancing of speaker levels.

Control Layout

At the left end of the front panel are five thin, bar-type pushbutton switches. The first three are interlocked "Mode" buttons for Dolby Surround (indicated with a double-D symbol), "Stereo," and "Mono." The next button, "Defeat," is tied with a line and a "Surround Synthesis" label to the "Stereo" and "Mono" buttons. "Defeat" does not affect the Dolby Surround mode, which, of course, is normally the best match for sources that have been specifically encoded for Dolby Surround. "Stereo" and "Mono" select synthesized surround sound to go with any stereo or mono sound source, unless "Defeat" is on. The last of the five buttons, "Tape Mon.," switches to recorder input connections.

In the center of the panel are two rotary "Input" controls ("Level" and "Balance") and a six-position "Digital Delay" rotary switch. Above "Level" is a horizontal, five-LED level meter. From the left, the first four LEDs are green; the last one is red, indicating maximum allowable input level. "Balance" has a single green LED above it which turns on when there is center/monaural energy and the control is set correctly for good Dolby Surround separation. The pot has no center detent, nor should there be: The control adjusts for out-of-balance sources within limits of ±9 dB.

The "Digital Delay" switch has positions for "16," "20," "24," "28," "32," and "36" mS. This is a very good range for delay, and I have commented before that processors with a 30-mS limit would be compromised in some large rooms. Delay settings do not have to be precise, and this decoder's 4-mS steps are quite acceptable. Further to the right are two rotary "Volume" controls, "Surround" and "Master." Above the latter pot is the red LED that shows when the remotely controlled "Mute" is on. To the right of the "Master" pot is the infrared "Remote" receptor, and above this receptor is a green LED that flashes to confirm reception of the remote's commands. All panel designations are in gold and are easy to read against the black panel. Large, gold index lines on the four rotary pots and the rotary delay switch make it easy to see these controls' settings from a considerable distance.

The matching Shure HTS amp custom-tailors itself for side, center, or subwoofer speakers in the context of the entire system.

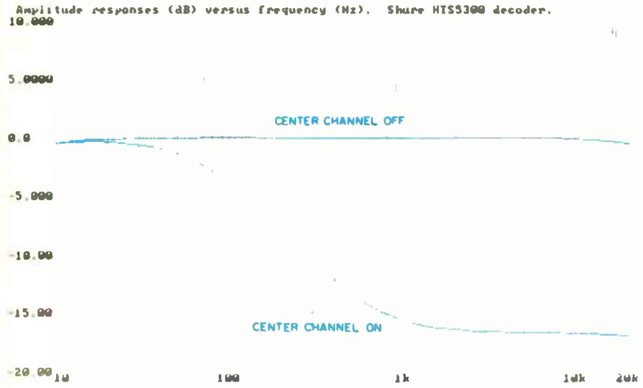


Fig. 2—Frequency response of main stereo channels with a mono input. The high-frequency reduction with the center-channel output on is deliberate; see text.

Further to the right, at the end of the panel, is the very useful Image Analyzer display, exclusive to the Shure HTS decoders. The display consists of shaped red LEDs that form a trapezoid. A center bar at the top illuminates when there is center-positioned energy. To the left and right are shoulder-shaped bars that turn on with left and/or right signals. Completing the figure are a rounded "L" at the bottom left and a backward rounded "L" at the bottom right. Both of these will turn on when the source has surround-type information. This display conveys immediately whether the source is strictly monaural, stereo, and/or has surround artifacts to be utilized. The varying intensity of each LED bar indicates the strength of each directional component of the signal.

The remote control is simple, having just "Master Volume," "Surround Volume," and "Mute" controls. The volume controls are long bars at an angle, which makes them easy to actuate when the control is held in the right hand. The bars are rockers: Pushing down on the grooved left ("−") end reduces volume, and pushing on the smooth right ("+") end increases it. A push of "Mute" will cut off all outputs or restore them; pushing either volume bar will also disable the mute. Actuation of any remote-control function illuminates a bright green LED near the transmitting end of the remote. If "Mute" is held in for 3 S, the HTS 5300 test generator is turned on. Then, a Noise Sequence circuit for speaker balancing automatically steps the generated test tone (from left to center to right to surround, and repeating) for adjusting levels as needed. Another push of "Mute" turns the sequence off.

Seven trim pots are available from underneath the unit. At the left front is "Mono Enhance," for modifying the factory-set mono enhancement if desired. Access is obtained near the back panel to the pots for "Front" ("L" and "R"), "Surround" ("L" and "R"), "Center," and "Subwoofer." Next to each access hole is an arrow indicating rotation direction to increase level. These trim pots can be very important if one or more amplifying channels lack any means of controlling volume.

On the back panel, from right to left, the first jack is for an optional "Wired Remote." Next is a pair of gold-plated stereo phono jacks for "Input," two pairs of "Tape" jacks labelled "Send (Record)" and "Return (Play)," "Outputs" jack pairs for "Front" and "Surround," and individual jacks for "Center Output" (top) and "Subwoofer Output" (bottom). A white line from the "Center Output" jack guides the user to a three-position slide switch ("Off," "Lo Cut," and "On"). It is important that this switch be set correctly because it affects how the signals are processed to the main speakers as well. Above this switch is a "Remote Sensor" jack for use with the optional remote-extender accessory, an infrared remote sensor that can be sited to pick up instructions from the remote control where the HTS 5300 itself would not be in the user's direct line of sight.

I removed the top and side cover to get a look at the inside construction. There were two large p.c. boards, one covering two-thirds of the chassis area and the other most of the remaining one-third. Support for the two boards was good, and they were less springy than I thought they would be. The power transformer mounted in the small space not used by the boards, was just warm to the touch after hours of operation. Immediately, I was impressed by the large number of quality components in a very orderly layout. There were a number of transistors as well as many ICs. Parts were all identified, and many of the trim pots were also labelled by function. Most pot adjustments were held in place with a spot of glue, helping to ensure long-term stability.

Most interconnections were made with multi-conductor cables, some with plugs and some soldered. I could not see the foil side of the boards, but my examination of component leads and holes on the top showed that solder flow was excellent. There was one fuse in clips. Because of its sheet-metal side rails, the chassis was quite rigid, even more so with the cover back in place.

The reader should be aware that the HTS 5300 does not have a power switch, though I do not see this as a potential problem for most users. If desired, the decoder can be plugged into a switched outlet on a preamp, integrated amp, or receiver.

Measurements

Let me first point out that all of the measurements were made *after* all of the listening and viewing.

Figure 2 shows main-channel frequency responses with a mono input. When the center-channel output was off, response was basically flat, down 0.1 dB at 20 Hz and 0.9 dB at 20 kHz. Output was down 3 dB at 3.1 Hz for both channels and at 30.0 and 39.0 kHz for left and right, respectively. When the center channel was on, the response of the main cnannels with the mono input was definitely far from flat. Note how its level, just about 0 dB at the lowest frequencies, falls off steadily with increasing frequency until reaching a shelf at about −17 dB for frequencies above 1 kHz. Briefly I was puzzled, but then, the light: When the center channel is on, it *should* be carrying the in-phase energy (especially the higher frequencies), and the stereo channels *should not*. This is one more example of the HTS system's automatic level and response compensation.

Shure's latest Acra-Vector logic decoder has 80% more sensing points than prior models, for smoother and more accurate imaging.

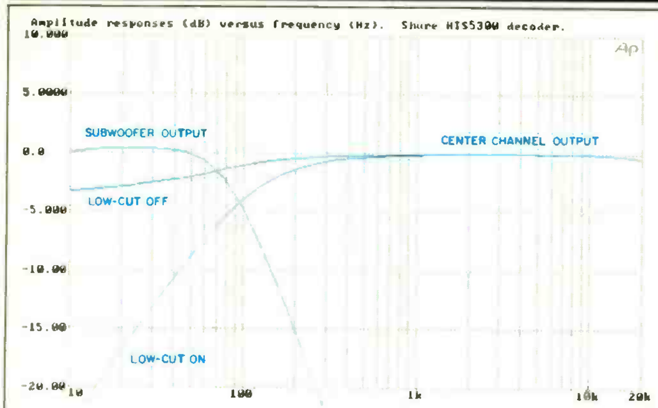


Fig. 3—Frequency response of center and subwoofer channels; see text.

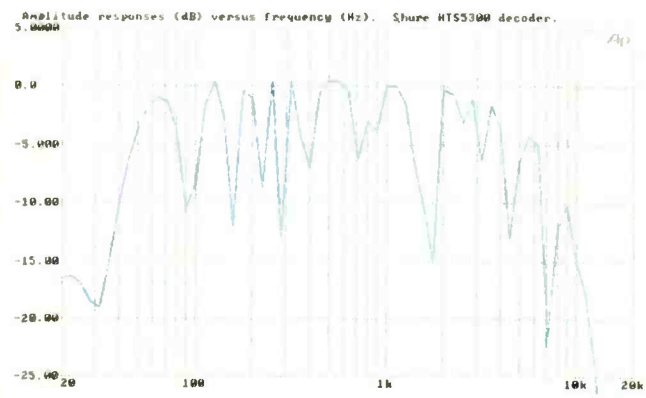


Fig. 4—Frequency response of surround channel, with left and right inputs of opposite polarity. The comb-filter effects shown are normal for such signals, but channel response is essentially flat, as seen by the envelope of the curve peaks.

Figure 3 shows the responses of the center and subwoofer channels. The response of the center channel with the rear-panel "Lo Cut" switch off was down 2.5 dB at 20 Hz and down 0.9 dB at 20 kHz; the droop at the lowest frequencies was purposeful, to make the total (center plus left and right) acoustical power flat with the Shure HTS speakers. This would be easy to equalize, if needed, with loudspeakers of other brands. The center-channel response is also shown with the rear-panel "Lo Cut" switch on. The roll-off below 200 Hz could be of benefit if a limited-response speaker is used for the center channel, particularly with a subwoofer. The response curve for the subwoofer channel shown in the figure has a roll-off above 80 Hz at a rate of 12 dB/octave. I could have trimmed the output down to the same maximum level as the other curves, but I didn't take the time to do that. The surround channels have no output unless nonidentical signals are fed to the right and left main inputs, and phase differences between these signals normally produce comb-filter effects. This is shown in Fig. 4, for which left and right input signals of opposite polarity were used. Frequency response can be roughly gauged from the envelope of the curve's peaks, but the apparent surround-channel response varies with the mix of signals in the main channels. After observing several such mixes, I'd say that surround-channel response is about 3 dB down at 40 Hz and 7 kHz.

Input sensitivity at 1 kHz was 250 mV for the maximum acceptable input level (the point at which the red LED of the level indicator just lights) and with the input-level control at maximum. Input clipping appeared at 3.9 V and output clipping at 4.9 V. The signal-to-noise ratio was 90.6 dBA for the main channels and 92.1 dBA for the surround channels, with a 1-V reference. Figure 5 shows the THD + N for the main channels, 0.04% or less across the entire band, at 1 V input and output. The surround-channel figures reached 0.06% over much of the band, but this is really quite good and well within specification.

The input impedance was 72 kilohms, and the output impedance was 5.4 kilohms. The input impedance is a good figure and was not affected by the setting of the input-level pot. The output impedance, however, would be on the high side if used with an amplifier having an input impedance of 10 kilohms or less. The Shure HTS 50SPA amplifier's input impedance is 100 kilohms, which is plenty high for the 5.4 kilohms of the decoder output. The two sections of the input-level pot tracked almost perfectly, staying within ±0.2 dB over its 20-dB range. The sections of the "Master" volume control tracked each other within 1 dB, from wide open to more than 80 dB of attenuation—*outstanding*.

A check of the output-level trims on the bottom panel revealed that each was factory-adjusted to its maximum setting and that close to 20-dB attenuation was possible with each. Exact Dolby Surround input balance with a mono input (null in the surround outputs) was achieved with the control at a little past 12 o'clock. The best null was close to 60 dB deep at 1 kHz, although the adjustment was touchy and the level bounced around. Typically, the nulls were 35 to 45 dB deep across the frequency band, which is very good. The separation between the main left and right channels was between 45 and 64 dB. (The lower figure was measured using a higher-than-normal level.) I tried a test videocassette that Shure had supplied. With a good level from the left-channel speaker, I heard substantially nothing from the right-channel speaker and a very low level from the surround speakers.

The delay adjustment range was from 16 to 36 mS in 4-mS steps. Each setting was accurate within 0.3 mS. The polarity was the same as the input at all channel outputs. The input-level meter's green LEDs turned on at −29, −18.8, −12, and −6 dB relative to the red LED turn-on at 0 dB. The red LED turned on with a 90-mS, 5-kHz tone burst when the continuous level was set 1 dB above turn-on. Decay time was about 230 mS for the bottom LED to just

The Image Analyzer display conveys at a glance if the source is monaural, plain stereo, or has usable surround characteristics.

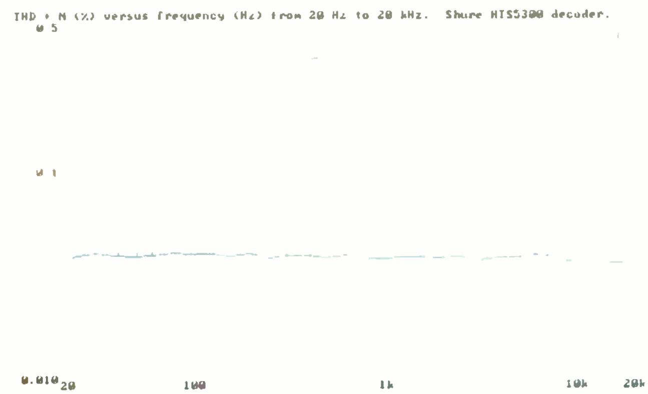


Fig. 5—THD + N for main channels.

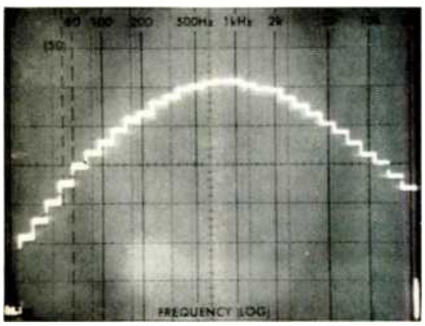


Fig. 6—Third-octave spectrum of speaker-balancing test signal generated by the Shure HTS 5300; see text.

Fig. 7—Listening setup used in evaluating the HTS 5300.

turn-off. Clipping, with a test tone, was 5.5 dB above the red LED's threshold. This simple meter will give good indications of level, although it does not respond to the shortest peaks. Shure recommends input level be set for just occasional red flickering, and this instruction should be followed.

Figure 6 shows the third-octave spectrum of the HTS 5300 test signal used for balancing speaker levels. The noise is broadband but is peaked in the middle of the band. This is actually good, because it minimizes sonic differences from speaker to speaker caused by response deviations at the frequency extremes.

Use and Listening Tests

The evaluation system, including the Shure HTS 5300, is shown in Fig. 7. Input and output connections were made in a jack field, which facilitated making a change to my reference Yamaha DSP-1 processor without too much delay. A Yamaha AVC-50 amplifier was used for input switching of the various sources: A Yamaha TX-900U AM/FM tuner, a Magnavox FD1041 CD player, a Sanyo VCR-7200 Beta VCR, an Akai VS-555U VHS VCR, and a Yamaha LV-X1 videodisc player. For power amplification, I used the second section of the AVC-50 for the main stereo channels, a Lafayette amplifier for the center channel, and two channels of a Yamaha four-channel M-35 for the surround channels. The speakers were two JBL 4301s (main stereo), a JBL 216 (center), a self-powered Triad Design HSW-300 (subwoofer), and two Dynaco A25s (surround). Because I used the Triad Design self-powered subwoofer, which has its own left/right bass summing network and crossover, I did not use the HTS 5300's subwoofer output. The Akai VCR was used as the stereo-TV decoder. A 26-inch Zenith TV was the video monitor.

The owner's manual concentrates on how to interface the HTS 5300 with the rest of the Shure HTS Theater Reference System, but many of the instructions are easily applied to other equipment. To make certain there is no confusion, the manual has a section on interfacing with other equipment, including cautions on making certain that polarity is correct. There are brief but lucid instructions on setting the delay time to match specific listening rooms, a short but helpful section on program sources, and a list of film releases that have been surround-encoded.

By naming the division responsible for surround products Home Theater Sound and by making "HTS" part of the model designations for these products, Shure emphasizes that home video/movie viewing is primarily what the system is designed for. Each year, more and more movies are released with Dolby Stereo encoding, which shows as Dolby Surround encoding on videodiscs and videocassettes for the home user. My viewing and listening concentrated on movies, but I also listened to CDs and FM stations.

For an X/Y display of the left/right input signal, I used an oscilloscope. I set the HTS 5300's delay at 24 mS to match my listening room. I confirmed the manual's statement that aiming of the remote control was noncritical. I even pointed the remote behind me and directly to the sides, and it worked reliably.

I tried a few stereo TV shows but found little of sonic interest. On CBS, *TV 101* had all of the dialog, even for off-

The peak in the middle of the broadband test signal makes it easier to balance speakers having different frequency response limits.

The complete HTS Theater Reference System would include the HTS 5300 decoder (shown here with remote control and remote extender), three of the HTS 50SPA amplifiers, four of the small speakers shown at the left, one center-channel speaker (middle), and one subwoofer (right).

screen action, right in the center. This sonic result was confirmed by the straight line at 45° on the 'scope and by the center-bar illumination in the decoder's panel display. The music and effects had some stereo spread but substantially no surround. When I watched a mono Celtics/Nets basketball game, the mono synthesized mode was best; my enjoyment increased after I raised the surround level to get to a good crowd-noise level. Overall, results for stereo TV with the Shure HTS 5300 were superior to those with the reference Yamaha DSP-1.

The first movie I tried was *Wall Street* (HBO simulcast), with Michael Douglas, Charlie Sheen, and Daryl Hannah. It was Dolby Surround encoded, and stereo spread in the music and effects was good. Surround information was just occasional (indicated better on the Shure HTS unit than on my 'scope), but it was used effectively. The dialog was strongly and, in the main, realistically centered. The 1987 movie, *The Whales of August*, with Lillian Gish and Bette Davis, was tried in the videocassette version. The sound was mono, but "Mono" synthesized surround did not improve the listening. On the other hand, a cable broadcast of *Jeremiah Johnson*, the 1972 mono-sound movie starring Robert Redford, was significantly improved with the same setting. Dialog was well centered, and surround effects were worthwhile.

When the rest of the family decided that they wanted to watch *Ben-Hur*, with Charlton Heston, on Showtime, I agreed reluctantly: What would be possible from an old 1959 movie? And it's so long! Unenthusiastically, I selected Dolby Surround and waited for confirmation that a synthesized mode would be needed. In a very short time, I realized that surround sound was alive and well and living in a 30-year-old movie. The dialog was clearly defined in position, on or off the TV screen, and there was even a shifting of sonic position between actors within the same scene. The 'scope showed the straight line for the monaural character of the talking but changed its tilt anywhere from straight up and down for all the way left (off screen to the left) to horizontal for all the way right.

Music and effects in *Ben-Hur* had continual stereo information, and the surround-sound quality allowed setting the level high without any detectable speaker localization. During the chariot race, the cheering by sections of the arena crowd for their respective heroes was positioned around the room. The storm after the crucifixion scene was very effective, especially the thunder—although it was somewhat distorted. The soundtrack had some other limitations, such as compression of the cymbal crashes in the music at the end of the movie. There were jumps in the positioning of the dialog, but the great majority of the time, the change in localization matched the change in the scene. The panning mixer missed the timing just a few times in a very long movie. Despite my initial skepticism, *Ben-Hur* gave an emphatic demonstration of what is possible with a good source and a good decoder.

I switched to videodiscs as sources and picked *Ladyhawke*, with Matthew Broderick, Rutger Hauer, and Michelle Pfeiffer (Warner Home Video). This is one of my favorites, and the sound quality is excellent. The sounds of Broderick's escape from prison right at the start of the movie were more detailed and had better clarity than I have noticed with any other system. Surround sound was very good throughout, both for music and effects. Dialog was very clear and was never spread in character. I would have preferred some shifting of dialog position to go with the picture, but the 'scope and analyzer displays showed that the source did not provide any such information. In a previous "Equipment Profile," I had commented on another system's popping in one part of a scene of *Ladyhawke* and suggested that the problem might have been with the videodisc. However, the HTS 5300 showed no such negative artifacts from beginning to end of the selfsame disc.

Back to the Future, with Michael J. Fox and Christopher Lloyd (MCA Home Video), delivered very good surround on the music and effects using the Dolby Surround setting. The skateboard chase and the car take-offs and landings were particularly good. Again, I would have preferred at least some panning of the dialog, but none was in the source.

For movies, the results with the HTS 5300 were noticeably superior to those with the DSP-1.

I then turned my attention to Compact Discs. *Carols from Winchester Cathedral*, with the Winchester Cathedral Choir directed by Martin Neary (ASV CD QS6011), had a fairly smooth sound field with stereo surround synthesis, but it was noticeably better with Dolby Surround. Bach's Brandenburg Concerto No. 1, from the I Musici set (Philips 412790-2 PH2), was slightly better with stereo surround synthesis than with Dolby Surround. Both were certainly superior to stereo without surround. Delay settings from 20 to 28 mS were all good for these two CDs; delays longer than 28 mS yielded a more spacious sound, but it was not as smooth.

Mozart's "Posthorn Serenade," performed by the Prague Chamber Orchestra with Sir Charles Mackerras (Telarc CD-

Some recordings of music fared best with the Dolby Surround setting, others with stereo surround, but all benefited.

80108), was best with Dolby Surround. "Tam O'Shanter" by Malcolm Arnold, from *Scottish Overtures* with the Scottish National Orchestra and Sir Alexander Gibson (Chandos CHAN 8379), seemed equally good using Dolby Surround or stereo surround synthesis. Some cymbal crashes were far better than they would have been with normal stereo. I wanted to make the surround sound more live (reverberant) with these two CDs, but there was no way to do that.

Time Warp, with Erich Kunzel and the Cincinnati Pops (Telarc CD-80106), produced strong surround indications on the HTS 5300 panel display. It wasn't surprising that Dolby Surround was a good choice for a number of the pieces. "Ascent," by Don Dorsey, was one of the best from this collection. Leonard Bernstein's *West Side Story* (Deutsche Grammophon 415254-2 GH2) has limited surround information, and the music remained too much front-centered no matter what I tried. Emmylou Harris' *The Ballad of Sally Rose* (Warner Bros. 25202-2) had good surround indications, and Dolby Surround was the preferred mode.

For the carols and the Bach Concerto, I had slight but firm preferences for the DSP-1 processor's "Chamber" program setting with adjusted reverberation. The HTS 5300 could not generate the liveness I wanted for the Mozart and Arnold works, although I could get it with the DSP-1's "Hall" programs. Various DSP-1 modes were also preferred for the *West Side Story* and Emmylou Harris CDs.

On FM broadcasts, the HTS 5300 kept vocals and announcements centered when I listened to rock music. It did not make announcements sound odd, as music-oriented reverberation systems do. In many cases, reverberation on announcements is quite acceptable, so I would be inclined to use such reverb systems on those classical recordings that would benefit from sound-field manipulations not possible with the Shure HTS unit.

The Shure HTS 5300 decoder provided the best localization of dialog and effects for movies, in any format, of all surround processors tested to date. Setting the level for good surround sound without distracting localization was less critical than it was with Shure's previous models. The HTS 5300 did not generate any spurious artifacts from any of my sources, as has occurred with other units. It provided very satisfying sound fields with certain CDs and FM music, but it was not a match for the reference Yamaha DSP-1, with most music, in generating realistic hall illusions.

The complete Shure HTS Theater Reference System offers possible advantages to the dedicated movie fan. Although the system's price is high, it is not necessary to buy all of its components, and the cost of the HTS 5300 is in the same price range as other decoders. If the prospective user's emphasis is on theater sound in the home, this Shure HTS processor should definitely be considered.

Howard A. Roberson

AD INDEX

The complete review:

A BEST-BUY BREAKTHROUGH OR THE START OF A NEW WAVE?

I am reluctant to call any given transistor power amp a "best buy" or breakthrough. From my talks with designers and other audiophiles, it is clear that the state of the art in power amplifiers is about to change. From where I stand, the Adcom GFA-555 is the first sample of this new wave. It is so clearly superior to past amplifiers in the low- to mid-priced range—not to mention most amplifiers two to three times its price—that I can unhesitatingly recommend it for even the most demanding high end system.

The GFA-555 does everything well, and most things exceptionally well. It provides superb, well-controlled bass with far better speaker load tolerance than most amps. Its midrange and treble are remarkably low in coloration. There is no hint of hardness, and none of the loss of inner detail common to transistor amplifiers.

"The Adcom's soundstage is sufficiently superior that even those who claim all power amplifiers sound alike might hear the difference."

With the exception of the Krells, I have never heard a more detailed, natural, and extended upper four octaves in a transistor amp. The Adcom may even be a legitimate rival to the Krell; it's brighter and more dynamic, and somewhat more open. And, like the Krell, it gives the impression, on really good material, that the amplifier simply isn't there, on really good material. Nor is the Adcom romantic or sweet, like New York Audio's new Moscodes. Rather, it offers natural upper octave detail that the latter miss. Other amplifiers have similar upper octave performance, but I unhesitatingly recommend the Adcom over the very stiff competition from Tandberg and Threshold.

The Adcoms' soundstage is sufficiently superior that even those who claim all power amplifiers sound alike might hear the difference. It comes very close to the better tube power amplifiers in providing detailed, stable, realistic imaging with natural depth. It is not an Audio Research D-250, but is extraordinarily holographic—I suspect almost embarrassingly so. This kind of soundstage has previously cost at least $2000.

I am also highly impressed with this amplifier's dynamics. Once again, it is not going to survive a one-on-one with the Audio Research D-250 or Conrad Johnson Premier Fives, but it rivals any transistor power amplifier in its power class that I have heard—including high-powered receivers or amps with trick power supplies—at any price. It provides these dynamics into virtually any load without bloat, restriction of sound, or change in timbre. For all the nonsense published by most manufacturers about driving complex loads, this amplifier actually delivers.

The Adcom does not lose sweetness and detail as its power goes up. I am normally leery of transistor amplifiers rated much above 100 watts; they too often blur detail and harmonic information, and this sonic price tag is far more costly than the added power is worth. This does not happen with the Adcom unless the distortion lights are blinking, and they only blink when the amp is delivering well over its rated 200 watts per channel (8 ohms) or 325 watts (4 ohms). By comparison, once-outstanding high power amplifiers like the Hafler DH-500 now sound annoyingly veiled.

With a minor dealer modification, you can even drive 1 ohm loads like the Scintilla. I can't measure whether the Adcom delivers its rated 800 watts per channel into 2 ohms, or 20 amps peak, but I *can* tell you that it does a superb job of driving this superb speaker. Anything in its price range (or even close) generally changes timbre and degenerates when driving the Scintilla at 1 ohm.

"For all the nonsense published by most manufacturers about driving complex loads, this amplifier actually delivers."

I'm going to have to say a few words about its technology before I give Adcom a swelled head. You'll be happy to note that the manufacturer claims for the GFA-555 a simple gain path, a 700 watt toroidal transformer, a well- regulated high current power supply, new ultra-stable bias circuitry, direct coupling, no current limiting, and no output inductor. More substantively, its harmonic shape mixes suitable yinyang while avoiding the curse of pyramidology. This, of course, means that it weighs 34 pounds, has simple rack-mount black styling, pilot lights, warning lights (to indicate distortion levels above 1%), and measures exactly 7 3/16" by 12 1/4" by 19".

More pragmatically, the technical specifications are significant in that they represent reasonable bandwidth (4-150,000 Hz), damping (150-200), gain (27 dB), and noise (-106 dB). Of these, only the noise specification is outstanding. No attempt is made to beat distortion records: .09% THD at rated power into 8 ohms, and .25% into 4. I have heard so many power amplifiers with infinitely (well, an order of magnitude) better specifications sound so much worse; this may be the amplifier whose sound could convince *Stereo Review*, *High Fidelity*, etc. that their present measurements are virtually worthless.

I suspect that the Adcom is going to force many designers in the $1000-1500 range to either make radical improvements in their products over the next six months, or look at the possibility of retiring from competition. This is a "must" amplifier to audition before you spring for anything close in price. If the Adcom is simply the first of a whole wave of good amplifiers, it will help revitalize the high end for the average audiophile, and force most manufacturers into more reasonable pricing. Now, Adcom, if you can only come up with a preamp as good! **AHC**

"I suspect that the Adcom is going to force many designers in the $1000-1500 range to either make radical improvements in their products...or look at the possibility of retiring from competition."

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Manufacturer's note: Approximate retail prices listed in order of mention in review:

Adcom GFA-555	$ 750
Krell	2300-7500
N.Y. Audio Moscode	900-1600
Tandberg	1000-2000
Threshold	1490-3150
Audio Research D-250 (MK II)	6000
Conrad Johnson Premier 5 (pair)	6000
"high powered receivers"	?
"amps with trick power supplies"	?
Hafler DH-500	850

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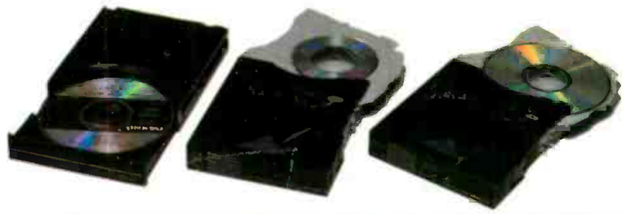

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