The Audio Critic®

Issue No. 23
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Do you need it? No. Do you want it? Yes!! Can you afford it? Hm... (See the Mcintosh reviews.)

Retail price: U.S. $7.50, Can. $8.50

In this issue:

We present an unusually comprehensive survey of Mcintosh stereo and home-theater components.

The long-promised in-depth coverage of FM technology, tuners, and indoor antennas begins, with a classic "tutorial" by Dr. Rich as the main feature.

An extended analysis of the different approaches to surround sound is followed by critical evaluations of multichannel audio-visual electronics.

We bring you our expected variety of loudspeaker reviews, in the price range from modest to insane.

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From the Editor/Publisher:

The elapsed time between this issue and the last one has been the longest in our history. Two relevant facts must be noted. One is obvious: this is, in effect, a double issue. I could have split it down the middle, made a few editorial adjustments and additions, and published it as two issues, twice as fast. You would have been charged twice. This way you 're getting (almost) two issues for the price of one. The other fact, which you couldn't have known, is that we were about to be plugged into a large, prosperous, and well-staffed publishing company, supposedly without losing any of our editorial autonomy but with all their forces mobilized to help us to publish on schedule. About six months were wasted before I faced the fact that they were just talking the talk but not walking the walk, and I was back to square one. The idea, however, was basically sound and should sooner or later be realizable with other partners. As I wrote last time, one way or the other, we 're here to stay.

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You may start your subscription with any issue, although we feel that new subscribers should have a few back issues to gain a better understanding of what *The Audio Critic* is all about. We still have Issues No. 11, 13, 14, and 16 through 22 in stock. Issues earlier than No. 11 are now out of print, as are No. 12 and No. 15. Please specify which issues you want (at $24 per four).

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Letters to the Editor

Our letters column is short this time, not because of a change in policy but as a result of preemption by other editorial material. We should be back to our usual six to eight pages next time. Letters printed here may or may not be excerpted at the discretion of the Editor. Ellipsis (...) indicates omission. Address all editorial correspondence to the Editor, The Audio Critic, P.O. Box 978, Quakertown, PA 18951-0978.

The Audio Critic:

For some reason I don't understand, when we attack a moral wrong, we usually focus on the supply side of the problem. For instance, we fault the prostitute and drug dealer more than the john (or jane) and drug user. And yet, if the market side dries up, the supply side usually follows suit.

In your ongoing war with *Stereophile* and *TAS* and the myths they preach, your focus is on the suppliers and supply of untruths, with the hope that as the audiophile reads this discourse, he or she will recognize the truth and will be set free. And you're to be commended for trying because sometimes it works.

But the greater truth is that the majority of audiophiles don't want to hear the truth. If someone suddenly were to discover a technology which was simple but had the capability of reproducing music indistinguishable from a live performance (by A/B blindfold comparison testing, of course), and particularly if this technology were inexpensive so that system price fell in the "consumer electronics" category, the majority of audiophiles would reject it outright. Why? Because it would eliminate their excuse to diddle endlessly with their systems and spend fortunes doing it. The majority of audiophiles don't want perfection. They want to be constantly striving for perfection. Their underlying motivation is not the enjoyment of music; it's playing with hardware.

A few years ago when I was an "audiophile" and subscribed to *Stereophile,* there was a letter in one issue from an audio salon dealer. He had done well in the business and had decided to rebuild his store, and to include in it a small but acoustically correct auditorium in which live performances could be presented for the enjoyment of his customers. After completion, what he found to his dismay was that he not only couldn't sell tickets to most of his customers, they wouldn't even attend for free.

What is this phenomenon? People have certain areas in which they excel. Men usually have an aptitude in mechanical and physical pursuits, and so excel in interests like stereo, ham radio, automobiles, sports, etc. The problem is that where we excel, we tend to overdo or become silly. And, as is well known, men get involved in their pursuits to the detriment of more important things—like their families. Women do the same, except their aptitudes and excesses usually fall in the areas of their homes, children, and mouths.

Harley, Atkinson, Olsher, Pearson, etc., may believe the stuff they write, or perhaps they know better but realize there's a bundle to be made by telling the "market" what they want to hear. Paul in his second letter to Timothy said, "For the time will come when men will not put up with sound doctrine [no pun intended, I'm sure]. Instead, to suit their own desires, they will gather around them a great number of teachers to say what their itching ears want to hear. They will turn their ears away from the truth and turn aside to myths" (II Tim. 4:3 & 4). He was talking about something far more important than music reproduction, of course, but the truth applies nevertheless.

Hartley Anderson
Long Beach, CA

P.S. I have no way of knowing this but am inclined to think that J. Gordon Holt, in his desire to publish regularly, reach a larger audience, and be less on a shoestring, wound up giving his publication to the Devil. As much as I get impatient for your next ever-late issue, please don't succumb to any temptations that might adulterate your publication.

I wish you were wrong about what drives the audiophile, but you're right.
—Ed.

The Audio Critic:

Can it be true? On page 11 of *The*

Audio Critic, No. 22, it's stated that power-output specification must take the form of "200 watts rms into 8 ohms..." and it's implied that this is required by the Federal Trade Commission.

It's the "must" that's surprising. I'm no longer surprised by "watts rms" in advertising or even in owner's manuals, but it is depressing to think that it might be embodied in a government requirement. Not that I'm surprised by government mistakes (it's much more surprising when the government gets something right), but somehow this doesn't seem like the kind of nonsense which would become official, considering the amount of participation by knowledgeable people from industry which must have gone into the creation of such a requirement.

Say it isn't so.

Sincerely,
Dick Sabroff
Lake Mills, WI

It probably isn't so, although I can't swear without looking up the regulation, and that shouldn't be necessary after the following explanation:

You are right, of course—there is no such thing, strictly speaking, as rms power. There is rms voltage and rms current, but voltage times current—which is power—cannot be expressed in rms form. I said "strictly speaking," but in the early days of audio we didn't speak strictly, and that's where my sloppy locution originated. I tend to lapse into it from time to time as my dotage approaches. The correct expression is "continuous power," and after some severe raps in the early 1970s from Paul Klipsch and other leading practitioners, the audio engineering community pretty much dropped the incorrect nomenclature. Whether the FTC originally used "rms" or "continuous" is academic at this point; the latter is correct, the former is wrong, you were right, I was careless, and that's that.

—Ed.

The Audio Critic:

...One thing that has always puzzled me about tweako reviews of electronic gear is the way in which a reviewer has the remarkable ability to hear the particular sonic signature of a component under review through all the presumed "colorations" imposed by the intervening equipment. For instance, how can a reviewer accurately characterize the sound of, say, a D/A converter if its signal must subsequently pass through two line-level interconnects, a preamplifier, a power amplifier, and speaker cable—all of which presumably superimpose their own sonic signatures as well—before reaching the speakers? One would think that there would be significant masking effects which would make reviewing an upstream component especially difficult. It would seem that the tweaks cannot have it both ways, with each component having a distinct, audible sound quality while one is able to isolate and characterize that particular sound quality through the sonic characters of all the other intervening components....

Sincerely,
Jeffrey B. Plies
Watertown, MA

When you enter cloud-cuckoo-land, the rules of clear reasoning are suspended. You somehow expect the fantasy world of the tweako reviewer to have some kind of internal logic—if my D/A converter sounds "dark" how can I be sure it isn't my preamp?—but that's not the way the tweako mind works. If the D/A converter was the last thing inserted into the system, it now dominates the sound!

—Ed.

The Audio Critic:

Your comment in Issue No. 22 about tweakos and the "audio performance" of light switches suggests you may have dealt with this matter somewhere in Issues 1-10, but just in case...

My experience in the Navy with some of our (then, 1970/80s) technology indicated that reliable electrical power sources made an enormous difference in the performance of our gear. As you may know, we used 440V/400-cycle well-regulated electrical power on shipboard. In port, we did not even try to keep our sensors and weapons aligned with each other because the supplied 117V/60-cycle power never reliably was that. Superficially, then, there does seem to be some merit in various tweaks such as better connectors (e.g., hospital-grade outlets), better house electrical-service grounding, maybe even better power cords, and so on. In these cases, ABX testing is mostly impractical. I don't think any such fixes can hurt, only unnecessarily impoverish. Is this an area you have covered or might consider covering?

Michael T. Corgan, Ph.D.
North Falmouth, MA

A few comments on power-line conditioners appear on p. 60 of Issue No. 16. There are residences (not too many) with grungy power coming out of the wall (low voltage, line noise, etc.); each case needs to be dealt with individually. Good outlets, good connector hardware, good plugs, good cords, good grounding, etc., prevent intermittents, assure unimpeded signal transmission, and hardly ever need replacement; they are highly recommended but have nothing to do with subtle audiophile-type differences in sound.

—Ed.

The Audio Critic:

Thank you for responding to my letter regarding the review of the Magneplanar MG-1.5/QR (Issue No. 21). Your comments are interesting and raise a number of questions worthy of further consideration.

...I had just begun to audition loudspeakers in the $1000-1500 range, and so I sought out the local Magnepan dealer in Connecticut (sorry, I'm not a Minnesota audio mafia member). After extensive auditioning and comparisons with a number of other highly regarded loudspeakers, I came to the conclusion that the Magneplanar MG-1.5/QR had superior midrange reproduction, a wide and deep soundstage, and superb transparency. Several competitors could reach lower in the bass and most were more sensitive, but none matched the overall performance of the MG-1.5/QR....

...I believe that an audiophile journal should aspire to the highest standards and that the loudspeaker reviews presently found in *The Audio Critic* fall short of these goals.

Yours truly,
Douglas M. Hughes, M.D.
Rochester, MN

It is with great reluctance that I excerpted your three-and-a-quarter-page letter so drastically, but there was no room here, and I felt obligated to acknowledge, even if inadequately, the continuation of the debate I had entered into.

Your letter is devoted mainly to insistent technical arguments, some of them repetitious and already answered by me last time, some of them new and valid, some of them new and invalid. It's all quite academic at this point because you are obviously determined to go to the barricades to defend your choice of the

(continued on page73)

In Your Ear


HONORABLE BOARD OF DIRECTORS, WE HAVE A PROBLEM.
OUR ENGINEERING DEPARTMENT HAS DEVELOPED THIS AMPLIFIER, NO LARGER THAN A PACK OF CIGARETTES. IT DELIVERS A MINIMUM OF *1000 WATTS* INTO ANY IMPEDANCE WITH LESS THAN *0.001%* DISTORTION AND IS EXTREMELY *COST-EFFECTIVE* TO MANUFACTURE.
THE PROBLEM IS THIS:
OUR AMERICAN AND BRITISH COMSULTANTS TELL US IT IS DEFICIENT IN *"RHYTHM AND PACE"* AND HAS INSUFFICIENT *"SLAM."*
IN YOUR OPINION, SHOULD WE *PRODUCE* THIS AMPLIFIER?
T.A.

Mcintosh Today (What the "High End" Was Originally Meant to Be)

By Peter Aczel
Editor and Publisher

Would you believe it? Very high-performance (and very pricey) audio equipment, scientifically engineered exactly as if the tweako cultists and their reviewer gurus didn't exist.

Imagine a team of high-end audio designers who are thoroughly up-to-date on their respective specialties (circuit design, electroacoustics, measurement methods, computer-aided technology, etc.) but have never heard of Robert Harley, Martin Colloms, Dick Olsher, Harry Pearson, Michael Fremer, and their ilk. All right, maybe they leaf through a tweako magazine every once in a while just for laughs but then go about their work exactly as they were taught in engineering school, totally uninfluenced by the tweako pundits. Refreshing, don't you think? Well, that's exactly how I see the current Mcintosh operation in Binghamton, New York, the slightly hicked-out "upstate" location, near the Pennsylvania state line, where they started in 1949 and never left. Never changed their basic audio philosophy, either, as I shall explain in a moment.

Meanwhile founder Frank Mcintosh and legendary CEO Gordon Gow are history; the company is entirely Japanese-owned (by Clarion, the car audio people); the President for the first four years of the Japanese reign was English; car audio products have been added to the line (predictably); yet the Mcintosh logo on an audio component means the same today as it did 46 years ago: the most serious kind of engineering without the slightest concession to trendiness, quality construction without showoff spending on meaningless overdesign, unique looks, and above all unproblematic operation under all conditions.

For example, a Mcintosh amplifier, preamp, tuner, or other electronic component will turn on and off without a sound out of the speakers—no thump, no pop, no sizzle, not even the tiniest transient. It was always so and still is. Even some very prestigious high-end brands are unable to make that claim. Not that it's the most important thing in the world, but such a Calvinist insistence on avoiding even the appearance of evil is highly characteristic of the company. Integrity is what they're into. That's what earned them their fanatically loyal dealer base, which has been the principal reason for their stability over the years. They could afford to ignore the whims and fetishes of the volatile audiophile market of the '70s and early '80s; they could afford to have quite minimal dealings with the tweako audiophile journals; their dealer network kept them afloat through thick and thin. The basic philosophy was that if a manufacturer is exceptionally supportive of the dealers and the dealers are exceptionally supportive of the customers, then a quality product is bound to sell whether it represents the latest-and-greatest trend or not. Yes, from time to time a certain stodginess crept into the product line as a consequence, but the present management appears to be capable of reconciling the "Mcintosh tradition" with advanced engineering. A recent visit to the Binghamton plant provided ample evidence of that.

Under one roof, which extends over workshops dating from 1949 and new wings with fully automated machinery, Mcintosh convincingly melds the old and the new in manufacturing operations. Here are glassworkers making the half-inch-thick glass plates for the retro front panels of Mcintosh amplifiers—they seem to exercise the craftsmanly care of old-time Venetian glassblowers but use the latest precision equipment. In another work area, computer geeks are polishing an incredibly sophisticated simulation program for multiple tweeters in line-source arrays. In yet another place, they are assembling just a few special-order antediluvian vacuum-tube power amplifiers with huge output transformers, wound in-house, for the nostalgia market. And over there, that's one of the largest and best-equipped anechoic chambers for loudspeaker measurement I have seen on either side of the Atlantic. Then there's the PC board facility... I could go on, but there's no end to the old/new contrasts. Had the new Japanese owners built a spanking new, 1990s-style plant, it wouldn't have nearly as much character as this organically evolved hodgepodge and couldn't possibly command the same kind of esprit de corps as currently prevails at Mcintosh. Being out in the boonies also helps; isolation from the alienated plastic industrial culture promotes clear, uncorrupted thinking, which is then reflected in the product line—at least as I see it.

That product line incorporates close to a hundred models today if one counts all categories; here I shall review just a few of the most representative examples. Please note that, as a departure from our usual editorial practice, these reviews do not appear within the articles devoted to other analog electronic components and other loudspeakers in this issue, although cross-references are included in those articles.

Stereo Power Amplifier
Mcintosh MC500

Mcintosh Laboratory, Inc., 2 Chambers Street, Binghamton, NY 13903-2699. Voice: (607) 723-3512. Fax: (607) 724-0549. MC500 stereo power amplifier, 500 watts per channel, with meters, $6500.00. Tested sample on loan from manufacturer.

This drop-dead, knockout power amplifier has the Mcintosh signature quality, probably more than any of the other Mcintosh products reviewed here. If an inanimate piece of electronics can ever be called sexy, this has to be the one. For once, I can actually see $6500's worth of good stuff built into the equipment—taking into consideration the dealer's markup, of course. Do you, as a discerning audiophile, *need* an MC500? No. Do you want one? Flash reaction: yes! After researching it and weighing the alternatives: maybe.

The Mcintosh MC500 is a solid block of huge passive components and active circuit assemblies neatly butted together, without even a sheet-metal cover, and it weighs (grunt!) 114 pounds after it's out of the shipping container. Two huge brushed-metal handles stick out of the front, but that's not what defines the MC500's look. Its principal mark of distinction is the pair of 5½-inch analog meters behind the half-inch-thick glass front panel. They are that special shade of Mcintosh blue and they stare right back at you as you watch the bouncing needles—but that's not the point; there are other power amps with big meters. Ah, but these are *genuine wattmeters,* not just voltmeters calibrated for watts into 8 ohms or 4 ohms. They actually measure the voltage *and* the current, multiply the two measurements, rectify and average the product, and display true watts with peak hold. Even without a world-class amplifier behind such an instrument, it would cost a mint. That's better value for your money than Wonder Caps, let me tell you.

The other major distinguishing mark that's immediately visible is the pair of large output transformers—no, no, autoformers is the correct term. Vacuum-tube amplifiers have output transformers, matching the high-impedance output stage to the low-impedance load. In the MC500, the *low*-impedance output stage is matched *more precisely* to the low-impedance load by means of the autoformer. In contrast to an output transformer, an autoformer is an exceedingly simple device, imposing virtually no signal-altering influences on the output signal path. It merely helps the power supply satisfy the current demands of any load closest to 8 or 4 or 2 ohms, as selected by the user. Instead of putting the money into a monster power supply which can deliver the same voltage into any load and thus double the power as the load is halved (that's the Krell approach), Mcintosh gives you the same high power into any load by means of impedance matching. The power supply is still very large and would undoubtedly do a decent job without the autoformer, but with the latter the amplifier becomes totally loudspeaker-friendly and has money left for other goodies (such as the wattmeters). I am not saying that this is "better" than the spendthrift brute-force approach, but it is equally good (at least for the vast majority of applications) and makes a lot of engineering sense—which is what Mcintosh is all about. In terms of bulletproof protection it makes even more engineering sense because shorts across the output of the autoformer are of no consequence whatsoever, even without all the other protection features of the amplifier. I'm a minimal-signal-path purist but I still like it! Only if the loudspeaker fluctuates over an improbably wide range of impedances—say from 1 ohm to 24 ohms or something crazy like that—will there be any doubt about the adequacy of autoformer coupling.

Aside from the wattmeters and autoformers, the MC500 operates pretty much like any other very high-powered amplifier. Both unbalanced (RCA) and balanced (XLR) inputs are available. David Rich's sidebar explains the circuit details. Build quality is outstandingly good, but don't expect exotic-brand capacitors and resistors on the circuit boards because Mcintosh engineers are into function, not fetish.

My measurements ran into a slight difficulty that proved to be purely academic in the end. On the Audio Precision test for THD + N versus level, the autoranging function of the instrument can generate transients that are apparently sensed by the Power Guard circuit (see the sidebar), which then activates prematurely. As a result, I was unable to obtain accurate THD + N curves up to full rated output and beyond. It made little difference, however, because (1) I was able to ascertain with less sophisticated instruments on my lab bench that continuous power output well in excess of 500 watts per channel was obtainable at each autoformer tap into the matching load, and because (2) the THD + N had dipped to -95 dB and even lower at the point where the test started to stumble, with every sign of moving in the same direction to even

Inside the Mcintosh Electronics

By David A. Rich, Ph.D.
Contributing Technical Editor

The following are my conclusions after taking off the cover of each unit, examining the PC boards and other component parts, walking through the circuit schematics in the service manual, and observing the unit in operation. The measurements and overall evaluations were left to the Editor.

Mcintosh MC7106

Nothing really interesting in the main circuit. The design makes no concessions at all to "audiophile" priorities and belief systems.

The input of each channel goes directly into the level-adjust pot. It is a very small control but it is sealed. The switch for the bridge operation is not as good, being a cheap open affair. A unity buffer formed by the NE5532N op-amp follows this. Electrolytics are used for dc blocking at the input and output of the circuit. Another electrolytic is in the ground leg of the feedback loop of the main amplifier (C_2 as schematized in Issue No. 20, p. 28, Fig. 6) to reduce the main amplifier's dc gain to unity. Complementary differential pairs follow. They are biased with a two-transistor current source and are resistively loaded. No degeneration is used. No follower stage is used. The second gain stage is a degenerated common-emitter amplifier that is dynamically cascoded. The dominant pole is set by a capacitor connected from the output of this gain stage to ground. The low distortion of the amplifier can probably be explained as due to cascodes in this stage. The only unusual topological feature of the amplifier is the connection of the collector of the differential pair that is not driving the second gain stage's base to the emitter of the cascode device. This folded cascode configuration increases the amplifier's open-loop gain somewhat. The output stage is a triple emitter follower. Three output devices are on each supply rail. The total transistor count exclusive of protection is 24 bipolar devices per channel.

The power supply for the entire amplifier consists of only one transformer, one diode bridge, and a pair of 36,000 µF capacitors. The capacitors are PC-board-mounted; thus a cheap strapping wire for the ground is not required. The Marantz MA500, costing about half as much per channel, is clearly going to have better channel separation, especially under load, given its separate power supply in each monoblock. The Mcintosh has a built-in relay for remote power-on through a proprietary control system.

Protection is extraordinarily comprehensive. The voltage difference of the main differential pair of the main amplifier is monitored. Under normal operation this is the summing junction and it should be very small. If the amplifier has reached it limits, then the difference voltage will get larger. This error signal is full-wave rectified and averaged. It then drives a light-dependent resistor (LDR). The resistive end of the LDR is connected back to the input of the unity-gain buffer IC described above. During normal operation the resistance of the LDR is high and it is effectively out of the circuit, although the tweako crowd would find its presence unacceptable. As the amplifier starts to become nonlinear, the resistance of the LDR drops and the input signal is attenuated. A light called Power Guard goes on when this function is activated. The output of the amplifier goes through a relay that activates for thermal shutdown and the presence of dc current on the output. The relay also is delayed in operation on power-on and opens on power-off. This prevents thumps and clicks at the speaker terminal. Traditional single-transistor overcurrent foldback protection is used and it is set to trip early. This circuit is responsible for the amplifier's unimpressive PowerCube. Power-supply inrush current limiters were said to be in this amplifier but not observed in the schematic.

A discrete "flash" A/D converter is used to form the LED voltage meter. The input to this is the speaker-terminal signal that has been full-wave rectified and averaged. As many as 66 op-amp sections plus a bunch of passive components and discrete semiconductors make for a very full board. Why cheaper monolithic ICs designed specifically for the purpose were not used is unclear. LED voltage level can be adjusted to change brightness, not something your average audiophile asks for.

Build quality is good but not ex-

lower distortion levels before clipping. That puts the MC500 into the uppermost category of power amplifiers in terms of low distortion, regardless of the vagaries of the Audio Precision. I saw virtually no difference in distortion at different frequencies, including 20 kHz, so that dynamic distortion can be said to be nonexistent.

Just in case the autoformer introduced any frequency aberrations, I wanted to make exceptionally careful and accurate frequency-response measurements. Using the unbalanced input and the 2-ohm output tap, at a level of 1 watt into the load, I measured -0.4 dB at 10 Hz, -0.1 dB at 20 Hz, 0.0 dB up to 20 kHz, +0.1 dB at 50 kHz, +0.25 dB at 100 kHz, and +0.05 dB at 200 kHz. Thus there is a quarter-dB "resonance" (if that's what it is) at 100 kHz and absolutely flat response in the audio range. The picture did not change as I moved the level up to 100 watts. Conclusion: forget about the autoformer as an extra circuit element affecting the frequency response.

The PowerCube test (see Issue No. 20, pp. 16-17, for a complete explanation and illustrations) proved to be illuminating. This is basically a test of current capability. Since the MC500, with its autoformer output, trades maximum voltage against maximum current—more V, less I at the 8Ω tap and more I, less V at the 2Ω tap than the active circuit's capability—it follows that the best possible results on this test will be obtainable at the 2Ω tap. The amplifier actually exceeded my expectations by drawing a truly nice PowerCube with that particular hookup. The cube had the predictable slight downward slant from the 8Ω tier to the 1Ω tier, but at every test traordinary. PC boards are double-sided. At the amplifier's price you get a fair but not very good deal.

Given the disappointing Power-Cube of the Mcintosh MC7106, I lean toward the Bryston alternatives, although six equivalent (>100 W) Bryston channels will cost a little more. The Bryston is a more advanced circuit design and equally well-engineered for a long life. The added protection in the Mcintosh adds belts to the Bryston's suspenders; for some this may add to a sense of security, but those seeking state-of-the-art electrical performance should consider Bryston. On a practical level, the Marantz MA500 monoblock amplifier is the device of choice for the vast majority of users looking for a >100-watt/channel multichannel system.

Mcintosh MC500

Basically the same circuit as the MC7106 but output devices per rail go from 3 to 10. Balanced inputs are provided. The op-amp that was the buffer in the MC7106 is used for this function. A turn-on delay muting circuit is added at the input. The dynamic biasing circuit for the cascodes in the second gain stage uses a current source instead of a resistor. There are separate bridge rectifiers and filter capacitors (22,000 µF) for each channel. The feedback loop is much more complex because of the autoformer. The main feedback loop is connected to the 2-ohm tap. The main amplifier drive point is somewhere between the 8-ohm and 4-ohm taps.

The true wattmeters are nice to watch, but a pair of the Bryston 7B-ST monoblocks (500 W per channel) will cost you $2605 less and give you comparable performance (and then some). Again we see the Mcintosh priorities. It is much harder to damage a power amplifier under a short or other fault condition if it has an autoformer, but we pay for this in cash as well as reduced output into a possibly mismatched speaker at a particular tap.

Mcintosh C39

Once you have a handle on the basic design of this highly elaborate AV control unit, you'll have no trouble understanding all the others, as they are all quite similar. (See the rest of the AV/surround equipment reviews elsewhere in this issue.)

The phono stage is the standard noninverting topology, with electrolytics used for C_1 and C_2 (see Issue No. 18, p. 18, Fig. 2). An NE5534AN is the active element. The problems with this topology were discussed in detail in Issue No. 18. A clever plug arrangement on an auxiliary input allows for one more line-level input (the 12th line input!!) if phono is unwanted. When a plug is put in this auxiliary input jack, the phono signal is removed from the bus and the auxiliary signal appears instead.

Input selection is through relays. The output of the relay block is buffered by a unity-gain inverting op-amp with electrolytics on the input and output for dc blocking. Mcintosh wants no pops or clicks under any circumstances, and as we shall see dc blocking capacitors appear throughout the circuit chain. The active element is a Motorola MC33178P dual op-amp, one section of which is used for each channel. Mono mix is engaged at the input of this stage with a relay.

The record bus is also selected through relays. All 12 inputs can be selected. A record processor can optionally be put in series with this bus. Plugging the processor in automatically opens the bus. The stereo bus signal is buffered with an MC33178P, again with input and output dc blocking capacitors. Four separate output jacks can be connected to the bus (VCR1, VCR2, Tape1, and Tape2). These are enabled by relays to prevent self-loops that could oscillate. The C39 has more flexibility for the serious tapist than any other preamp known to me, with 12 inputs and four tape monitor loops.

Area B (remote room) outputs are taken directly from the record bus. Each channel is sent to an electronic volume control made from a Micro Power Systems MP7529 dual 8-bit multiplying DAC (the part is a second source of the Analog Devices AD7528, with improved distortion specifications) and the two sections of the MC33178P. The MP7529 is a better part than the Sanyo and Toshiba devices that we have seen used by B&K, Onkyo, and Marantz. Blocking capacitors are at the input and output of the circuit. After going through a bipolar-based mute circuit (everywhere else this is done with a relay—do not ask me why the difference here), the signal passes to the RCA plugs.

Video and S-Video switching are identical. CMOS switches are used that are then buffered by a two-transistor discrete circuit before being routed to the output jacks. The unit has six video inputs and four outputs for the video section. The outputs are for two TV sets and two VCRs. A video modulator is also included for TV sets that do not

point with a reactive load the voltage was higher than into a pure resistance, the way it should be. Into a perfectly matched 2Ω resistive load, the dynamic power was 812 W (i.e., 40.3 V). At no test point was the output lower than 35.7 V or higher than 44.5 V. The absolute current limits of the amplifier do not emerge from this test, but proper execution of the design concept is evident.

Regular readers of ***The Audio Critic*** no longer expect me to discuss the "sound" of a properly designed amplifier, but for the sake of newcomers let me repeat for the nth time that such an amplifier has no sonic signature. As for value, other mega-amplifiers of comparably low distortion and high reliability can be had for less money than the Mcintosh MC500, but they will not have its almost redundant set of fail-safe protection features.

Multichannel Power Amplifier
Mcintosh MC7106

Mcintosh Laboratory, Inc., 2 Chambers Street, Binghamton, NY 13903-2699. Voice: (607) 723-3512. Fax: (607) 724-0549. MC7106 six-channel power amplifier, $3500.00. Tested sample on loan from manufacturer.

Take one channel of the MC500, reduce the power a great deal, take away the autoformer and the wattmeter, mutiply by six, and you have in essence the Mcintosh MC7106. It weighs less than half as much (53 pounds) and has a knobless, smooth, glass front panel. David Rich's sidebar explains the design in full detail, so I'll go straight to my measurements.

have video inputs. One assumes this is envisioned for a TV in the B zone, since it is hard to imagine a main system controlled by the C39 with a TV that does not have direct video inputs. The audio for the modulator can be selected from the Listen (A) or Record (B) bus. A pass-through for cable TV is provided, so cable signals can go directly to the TV.

The main audio signal is routed or bypassed from the Dolby Pro Logic decoder by CMOS switches. In Mcintosh designs, costlier relays are used at any location where the user can introduce a fault condition, such as the input or output jacks. A fault condition that takes the signal outside the supply rail will cause a CMOS switch to latch up. To eliminate this condition the power supply must be cycled, and in a worst-case situation the CMOS switch could be destroyed. Mcintosh does use cost-saving CMOS switches internally in the C39, where it is not possible for the switch to see an over- or undervoltage.

The Dolby Pro Logic decoder is the Analog Devices SSM-2125, and the CMOS switches are the Analog Devices SSM2404. The SSM-2125 is surrounded by numerous passive components required by the part. In addition, an op-amp-based fourth-order bandpass filter is included for the noise source used for setting up the Dolby Pro Logic decoder. The Mcintosh C39 has nothing out of the ordinary in this functional block.

The signals to the Dolby Pro Logic decoder are ac-coupled with electrolytics. The subwoofer channel is formed by summing the left, right, and (if the decoder is activated) center channel. This signal is filtered by a fourth-order (cascade of two identical second-order sections) lowpass filter. All filter sections in the C39, including this one, are formed with the Sallen-and-Key circuit. When this subwoofer signal is activated, CMOS switches put second-order highpass filters in the left-, right-, and center-channel signal paths. Again, sections of the MC33178P are used as the active elements for the filters, with electrolytics at the front and back for dc blocking. Why it is expected that one corner frequency and a fixed set of filter orders will work with all main-speaker/subwoofer combinations is beyond me, but this appears to be standard practice in AV land.

The surround output (S) of the Dolby Pro Logic decoder or the L+R+C signal (the choice depends on the surround mode and is made by CMOS switches) is bandpass-filtered (7 kHz highpass limit) and then sent to an LV1000N Dolby Pro Logic Delay Unit used by almost everybody. This device has an A/D, a D/A, and a digital delay circuit. Dolby B is built into the LV1000N and is used as a single-ended noise-reduction processor for the time-delay block as required for Dolby Pro Logic. This bandlimited mono signal can be used directly or it can be sent to the optional THX module. This DSP-based unit, for which we have no schematic, creates stereo rear channels. It uses a DSP chip and thus requires A/D and D/A converters. The THX block also takes in the left, right, and center channels. What comes out is left, right, center, and stereo surround. Holman equalization may also be applied to the front channels, but this information is not available. The subwoofer output bypasses the THX board. After all this processing I do not find myself wanting to rush out and purchase surround-sound speakers. *[Be tolerant, folks, of the village atheist.—Ed.]*

The good news is that true discrete 5.1-channel surround sound makes this whole board obsolete. The other good news is that the whole section of active electronics is bypassed by the CMOS switches when in stereo mode. The bad news is, if you really want Dolby Pro Logic or THX (why?), then the analog approach used here is not the best. What you want to do is digitize the incoming stereo channels with high-quality A/D converters like those found in DAT players, or take digital data directly off the laser disc or CD and have *all* the surround-sound signal processing done in the digital domain. After *all* the processing is done, digital data for the 5+1 channels is converted back to analog, using D/A converters of the quality found in good CD players. Now Onkyo does just this and gives you a whole integrated receiver (seven—count 'em, 7—power amplifiers, including three 110-watt/channel discrete units, an on-screen TV display, and a tuner) for $1880.00, so why do we not get DSP in the C39?

The only downside to this digital approach is that the peak level of the analog signals must be set not to overload the A/D converter. On the other hand, if the signal level is too low in amplitude, high noise levels will result because the full range of converter gain is not being used. For digital signal inputs there is no downside at all.

Once the 5+1 signals leave the surround-processing section, they travel to a three-stage volume control. Three stages are necessary to get the

The Mcintosh insistence on low distortion is clearly in evidence here too. THD + N with an 8-ohm load and a 20 Hz or 1 kHz input reached a minimum of -103 dB before the onset of clipping, which occurred at approximately 150 watts. The 20 kHz THD + N bottomed out at -96 dB; call it dynamic distortion, but who cares when the numbers are that good? With a 4-ohm load the minima were -103 dB at 20 Hz, -100 dB at 1 kHz, and -94 dB at 20 kHz; clipping occurred at 300 watts. All of these figures were obtained with only one channel driven. With two channels bridged and a load of 8 ohms, the minima were -102 dB at 20 Hz, -100 dB at 1 kHz, and -87 dB at 20 kHz; the clipping point appeared to be just a hair under 500 watts. Dynamic distortion is unarguable in the bridged mode, since the 20 kHz curve started to part company with the 20 Hz and 1 kHz curves at 1 watt (-80 dB). Thus a bridged MC7106 is almost the stripped-down equal of an MC500 at half price, with two channels left over—but not if you look at the finer points of performance and, especially, at the PowerCube.

The MC7106 did well on the PowerCube test only at 8Ω and 4Ω there was severe current limiting at 2Ω; especially into ±60°, and total collapse at 1Ω. Dynamic output voltage was 38.8 V (188 W) into 8Ω/0° and 36.5 V (333 W) into 4Ω/0°, with slightly higher output into all 8Ω and 4Ω reactive loads. Into 2Ω/0° the output dropped to 32.3 V (521 W), rose slightly into ±30°, and then went kaputt at the remaining test points Dynamic headroom appeared to be nil when checked against the continuous power output curves.

required range and resolution. Each stage is formed with one half of an MP7529BJN multiplying DAC and one half of an MC33178P op-amp, with the exception of the volume control for the left and right channels, which gets the lower-noise 5532N. The 5532N devices are used when low noise is required. They are not used all the time because of their high power consumption, according to Mcintosh. The first MDAC is set to have twice the step-height gain of the remaining two MDACs. In the process of flowing through this electronic volume control, the signal has seen three more electrolytics for dc blocking. A microcontroller is on the surround-sound board to control all the Dolby Pro Logic electronics and the three-stage volume control. One interesting thing is that the volume-control knob is not connected to a rotary shaft encoder. Instead, the volume control generates a dc level dependent on the position of the knob. This presumably allows you to use volume-control position as a guide to the system's level. Since the C39 has a nifty LED digital readout of level, this may appear redundant—but wait, the knob has no index mark! Figure that one out. The said dc level tells the micro what data to send to the multiplying DACs. The C39 even puts a motor on the pot to implement remote operation, instead of just changing the digital control signals to the electronic volume control directly. Different levels between the main and surround channels require the use of keypad controls (ugh). The microprocessor sends different data to the multiplying DACs as the keypad switches are depressed.

At this point Mcintosh must have stopped designing and grabbed some old boards from the parts bin. The L, R, C, and subwoofer signals are routed to an analog board with a loudness circuit. NE5532 op-amps are the active stage, one section per channel, and an unsealed Alps pot does the honors for control. They have another dc-blocking cap on the board also. No, there is no bypass for this board at all. Even stranger is the fact that L and R go into an analog balance control, another unsealed Alps pot. Now, the balance function could be implemented with the multiplying DACs used for the volume control. Software would then mimic the balance control as in other AV products, so what's the pot doing here? Other than creating a reliability problem, I have no idea. I do know one thing: you cannot adjust this control with the remote—and it is the one control function you would actually want to run on the remote!

And then the L, R, C, and subwoofer signals go to (are you ready for this?) the nondefeatable tone control block!?! Each pair of signals out of the four uses an NE5532 for the active summing block and an MC33178P for the bass control circuit. You get another dc-blocking electrolytic cap also. Does this circuit block and the loudness circuit block affect the frequency response of the C39, and will the situation get worse as the unsealed Alps pots age? Yes and yes. Why did Mcintosh do something so wrongheaded? Do not ask me. Simple CMOS defeat switches would have made the whole issue moot, but they are missing on the C39.

After passing through muting relays, the 5+1 signals appear on the RCA jacks. With the surround-sound decoder and subwoofer filter defeated, we have passed through six active stages and seven blocking caps to get to the Listen (Area A) unbalanced line output. L and R are also available as balanced outputs. The dual MC33178P is used for the single-ended-to-balanced converter of each channel, and yes, there is another dc-blocking capacitor at the circuit's output. Another MC33178P is used for the headphone amplifier. The center channel, if active, is summed into both channels just before the (you guessed it) dc-blocking capacitor.

As you would expect, a lot of digital logic controls the whole thing, including separate microcontrollers for the Listen (A) bus and Record (B) bus. Another microcontroller is used to control the hardwired CD and tuner remote links. That brings the total to four. Power for the C39 comes from two transformers. But it is no audiophile special. There are 3300 (µF capacitors on the unregulated rails. And the regulated rails are set to ±12 V by one pair of 7812/79M12 regulators. That's right, all the electronics discussed above run on just this pair of regulators. The digital electronics, of course, have their own regulated supply.

Opening the cover of the large chassis shows where the money went. At the back of the unit the PC boards are stacked four levels deep. In the front they are only two layers deep. The downside of all the boards is that the wiring harness is quite complex and with its interconnectors represents a reliability problem.

Plastic knobs and a front-panel plastic door are not what you expect at this price. 0

David Rich makes a techie wisecrack about the channel separation in his sidebar, but what I measured was nothing worse than 52 dB (at 20 kHz) and went as high as 98 dB (at 200 Hz). Furthermore, the noise floor of the amplifier is very low, -100 dB or better at all frequencies across the audio band (e.g., -112 dB at 1 kHz), as referenced to a level of 1 watt into 8 ohms.

Please read my comment in the MC500 review above anent sonic characteristics. In actual hands-on use the MC7106 behaved as agreeably as the MC500. I share David Rich's opinion that from a purely practical point of view six Marantz MA500 monoblocks, costing $1800, will do the same job in a home theater system as an MC7106 at almost twice that price. But there will be those for whom a Japanese amplifier can never be a Mcintosh.

Multichannel AV Control Unit

Mcintosh C39

Mcintosh Laboratory, Inc., 2 Chambers Street, Binghamton, NY 13903-2699. Voice: (607) 723-3512. Fax: (607) 724-0549. C39 Audio/Video Control Center, $3500.00. THX-M optional THX processing card for C39, $500.00. Tested sample on loan from manufacturer.

This is such an incredibly complex and feature-laden preamplifier and multichannel control center that it's much easier to tell you what *isn't* in it than what *is.* (David Rich in his sidebar attempts to cover the latter question.) Well, there is no provision for on-screen displays of any kind (but the front-panel knobs pretty much tell it all), no AC-3 (but that's very new, and very few have it), no Ambisonics (but only Nimbus CD collectors will miss it), no moving-coil phono (but who needs it in a home theater system?), and in general no signal processing in the digital domain a la Lexicon or Onkyo. So you could say this is a somewhat retro piece of gear, but what an impressive monster!

The quality of construction is what you would expect of Mcintosh; everything is rock solid and beautifully put together, but again don't look for insanely costly "audiophile-brand" capacitors or resistors where they are not needed because Mcintosh engineers see no point in that. The glass front panel is very handsome, but that's where all the knobs are, so it's always covered with highly visible fingerprints. Hooking up a home-theater system to the inputs in the rear is not as daunting a task as the complexity of the unit would suggest because everything is logically laid out and intelligently labeled. I left the nitty-gritty details to David Rich.

On the lab bench I was unable to duplicate some of the specifications published in the manual. The frequency response at full gain through a line-level input and Area A front output had a 0.35 dB bump at around 23 Hz with the unbypassed bass control at its detent; the specs don't admit that. Through the same signal path, at full gain, THD + N was -79 dB regardless of frequency at 2 V out and climbed to -71 dB before the onset of clipping at just over 6 V. The specs say I should have measured -86 dB or better but don't say at what gain or output. It is possible, since the volume control is much further downstream in the signal path than in typical stereo preamps, that I was amplifying noise from the early stages at full gain, but the 8 dB rise in THD + N from 2 V to 6 V does not indicate a noise-dominated measurement. Again through the same signal path, at full gain and 2 V out, left/right crosstalk was -54 dB at nearly all frequencies, rising to -48 dB at 20 Hz. That's OK but far from great. Better results would surely be obtainable if the bass, treble, and balance controls were bypassed or digitally implemented.

The phono performance was disappointing. With Phono in, Tapel out, 2 mV input, 40 dB gain, the RIAA equalization error was +0.34/-0.55 dB (worst case). The largest error was at 20 Hz, but the entire response curve was skewed. Frequency response deviations of that magnitude are audible. Through that same signal path the THD + N reached a minimum of -75 dB at 6 V out, at all frequencies, this time completely noise-dominated. The simpler phono signal path yielded better crosstalk figures: between -70 and -80 dB at any frequency above 500 Hz, with a worst-case reading of -48 dB at the 60 Hz hum frequency (2 mV in, RIAA-equalized).

Using the C39 was a pleasant experience, although the lack of on-screen displays creates minor learning-curve problems for your average remote-control jockey. Even so, there is a reassuring feel of solid quality to this control unit as you live with it and operate it.

AM/FM Tuner
Mcintosh MR7084

Mcintosh Laboratory, Inc., 2 Chambers Street, Binghamton, NY 13903-2699. Voice: (607) 723-3512. Fax: (607) 724-0549. MR7084 AM/FM stereo tuner, $1500.00. Tested sample on loan from manufacturer.

This quite new tuner arrived with a considerable delay behind all the other Mcintosh pieces reviewed here and was not measured in time for David Rich's FM article and tuner reviews in this issue. The following early impressions are based on a look-see under the chassis cover, a circuit schematic as interpreted by David, and a brief stint of the MR7084 in my reference system.

Mcintosh tuners of the past have been RF masterpieces (e.g., the fabled MR78, still a big item on the used-equipment market), but it would appear that the MR7084 isn't one of them. Indeed, I'm inclined to think —although I have no proof, only circumstantial evidence—that this is the exception to the generalization I made above regarding Clarion's noninterference with Mcintosh product design. I somehow suspect that Larry Fish, Mcintosh's veteran chief engineer and one of the best RF men in the business, was told by the front office that he must use some existing circuit boards, possibly from the (gulp) car audio line, to lower the cost of tooling up for the new model. If I'm wrong, I apologize in advance, but here are the clues:

The MR7084 is a $1500 tuner, yet it uses the Sanyo LA3401 multiplex decoder IC, which is spec'd by Sanyo "for stereo systems and portable hi-fis," instead of the top-of-the-line Sanyo LA3450, spec'd "for high-quality stereo systems" and better by at least a factor of two in noise, distortion, channel separation, etc. The LA3401 requires a notch filter because it lacks a pilot-tone canceler; the LA3450 incorporates the pilot-tone canceler, which is the preferable solution. Now why would Mcintosh choose the less good chip, at a negligible saving, for their expensive tuner? The chassis layout reveals the answer. Inside the handsome box there's mostly air; the actual tuner is crammed into a tiny circuit board. The LA3450 has a larger footprint than the LA3401 and requires more external components—it wouldn't fit on that weensy board, which couldn't possibly have been designed for the MR7084 from scratch, as I see it, but borrowed from a preexisting design.

The Sanyo LA1 175, which forms the mixer, VCO, 1st IF, and AGC, is another obvious space saver; these stages are discrete on other expensive tuners. Sanyo lists this particular IC as designed "for car radios, stereo system." Nor is there room on that tiny board for two sets of filters in the IF strip, so you get no wide/narrow selection on the front panel. Very high quality is claimed for the single set of three ceramic filters. The quadrature detector used for the FM demodulator, on the other hand, is not of high quality; it's part of the Sanyo LA1235 chip that everybody uses, but the best tuners don't use the detector part. Nor do the best tuners use the signal meter part; they have separate circuits for that, whereas the Mcintosh just uses what's on the LA1235.

Overall, the Mcintosh MR7084 is still much better built than the equivalent cheaper Japanese tuners; the PC boards are double-sided, and the parts quality is generally higher, but deluxe details and packaging are no substitute for advanced design. In actual use, the FM reception and typical signal quality at my location were satisfactory but strictly average; I much preferred, for example, the Rotel RHT-10 at exactly the same price (see David Rich's review). Measurements of the Mcintosh and further evaluation will be published in the next issue.

Loudspeaker System
Mcintosh XRT24

Mcintosh Laboratory, Inc., 2 Chambers Street, Binghamton, NY 13903-2699. Voice: (607) 723-3512. Fax: (607) 724-0549. XRT24 floor-standing 3-way column loudspeaker system, $7500.00 the pair. Tested samples on loan from manufacturer.

I have a problem positioning this basically very well-engineered speaker system against others in its category for a fair review, perhaps because there are no others in its category. Is there another speaker between, say, $6000 and $9000 that uses a large number of tweeters in a line array? I'm not aware of one. Other than Infinity's older IRS models with their zillion vertically stacked EMIT-type tweeters and the newer Genesis five-figure oil-sheik models by the same designer, Mcintosh appears to be the standard-bearer of the whole concept.

Indeed, I'm almost certain that Mcintosh designer David L. Smith's marching orders were not to create the best $7500 speaker he knew how to design, period, but the best $7500 speaker using a discrete-element line array, his specialty. (He is the author of a 1995 AES paper on the subject.) There were already two older, larger, and much costlier speakers in the Mcintosh line based on the same design principle, but they were—shall we say— slightly on the dinosaurian side, indicating a necessity for a more up-to-date, cost-effective, and polished design. That is exactly what the XRT24 is, and as such it must be called a complete success. If I needed a new speaker and had $7500 to spend, would the XRT24 be my choice? Frankly, no, but not because of the engineering quality, which is the usual problem. My reasons will be apparent as we proceed.

A pair of XRT24's incorporates 32 tweeters of a high-quality 1-inch aluminum-dome type, 16 per side, deployed in a vertical line above the woofer/midrange enclosure. The idea is to synthesize a pair of line sources, with the added benefit of tremendous power-handling capability. Now, 16 discrete elements in a line array, closely butted together and driven equally, do not oblige the designer by producing a smooth cylindrical wavefront in accordance with the idealized model of a line source. *Au contraire,* they yield an unacceptable radiation pattern with very ugly lobes. The drive has to be "tapered;" i.e., each element or group of elements has to be driven differently with respect to level and frequency range to achieve the desired summed response. I have personally seen and observed in action the highly ingenious proprietary computer program Mcintosh has developed to calculate the most effective tapering. The results are excellent in terms of summed response; the trouble is that 32 tweeters still possess some of the inherent limitations of a single tweeter. It would undoubtedly be affordable to choose the world's best tweeter and put two of them into a $7500 pair of conventional speakers, but 32 of them? As I said above, the tweeters are of high quality, but they aren't among the world's best. The Win (proprietary) and the Accuton (OEM), to name only two, are among the best; 32 such tweeters would raise the in-house parts-and-labor cost of a pair of XRT24's to more than $7500 without any other changes, never mind the wholesale and retail prices. I'll come to the sonic consequences in a moment.

The lower part of each speaker consists of a forward-firing 8-inch midrange unit and two rearward-firing 10-inch woofers in a sealed cabinet. The whole schmear, with tweeter column, is 7 feet tall but only 15 inches wide and 18 inches deep at the base. The occupied airspace is quite modest from the waist up owing to the slender tweeter column, which tapers from 10 inches wide at the bottom to 6 inches at the top. Each speaker weighs about 130 pounds. Since the high-frequency drive is shared by 16 tweeters, they can be crossed over at a relatively low 1.5 kHz; the midrange covers 250 Hz to 1.5 kHz; the woofers come in below 250 Hz. The elaborately controlled drive to the tweeter column is accomplished with a passive network; one amplifier channel per side is all you need.

I measured the frequency response of the XRT24 from 300 Hz to 20 kHz with the quasi-anechoic MLS method, positioning the calibrated microphone at a distance of 2.5 meters from the line array and at a height of 1.3 meters. That's not really a standard setup, but then the almost ceiling-high XRT24 is far from a standard loudspeaker, and I wanted to obtain meaningful results at a realistic distance and some kind of realistic average ear position. The resulting response on axis turned out to be quite excellent: ±3 dB from 300 Hz to 18 kHz, and that doesn't even tell the complete story because from 3.5 to 12 kHz the response was closer to ±1 dB, and if it had not been for a marked dip at 3 kHz, the overall response would have been pretty much within ±1.75 dB. Going 30° off axis horizontally—and that's generally the stereo listening position—actually smoothed out the response slightly, making it ±2 dB from 300 Hz to 12 kHz. Above

12 or 13 kHz the curve was a little rough regardless of the microphone position, but this was due to interference patterns, not tweeter misbehavior. I checked for tweeter ringing with tone bursts of different frequencies and found little or nothing, except at that 3 kHz dip, where there appeared to be a slight storage problem, nothing major.

The dual woofer, measured with the nearfield method, exhibited a more or less classic sealed-box response, declining at 12 dB per octave below a -3 dB point *(f_3)* of approximately 40 Hz. Since the fundamental resonance of the bass system, as indicated by its impedance curve, appears to be in the neighborhood of 32 Hz, it would seem that the response is slightly overdamped, maybe because the rearward-firing design assumes some boost from the proximity of the rear wall (12 to 2 inches suggested by Mcintosh for initial placement). Reservation No. 1: When I'm asked to spend $7500 for a loudspeaker system, I expect more or less flat response down to the limits of audibility, not a missing bottom octave. For example, a pair of Velodyne DF-661 three-way speakers, coupled with a pair of Velodyne F-1500R subwoofers, would cost you $4885 (in black) and give you 16 Hz to 20 kHz response with ultralow distortion. No, they wouldn't look nearly as impressive in your listening room, doctor, as the XRT24's.

Speaking of distortion, the XRT24 is actually very good in that respect, though no Velodyne. At a 1-meter SPL of 95 dB (normalized to 50 Hz), the nearfield THD + N of the woofer rose gradually from 0.4% at 200 Hz to 3% at 20 Hz, staying under 1% down to 45 Hz. At 90 dB the rise was from 0.3% to 2%, staying under 1% down to 40 Hz. Respectable figures, all in all. Midrange THD + N, at a 1 meter SPL of 95 dB (normalized to 600 Hz), fluctuated between 0.23% and 0.73%, with no clear relation to frequency; at 90 dB the readings were barely lower, between 0.2% and 0.6%. Again, not bad at all. The distortion of the tweeter array was hard to measure because the drive is not the same to all 16 units. After a lot of messing and some educated guessing I determined that the distortion generally stays in the 0.13% to 0.4% range at 90 to 95 dB. Thus I can't really fault the XRT24 on distortion; these measurements fall barely short of the Snell Type A, for example. I have already mentioned the tone-burst response of the tweeter array; the woofers and the midrange driver gave no evidence of storage when tested with tone bursts.

The impedance curve of the speaker is a roller coaster; the magnitude fluctuates between 2.6 ohms and 16 ohms, the phase between -50° and +45°. A power amplifier with a good PowerCube performance (i.e., load-insensitive) is highly advisable. Nominal impedance is 4 ohms. Separate low, mid, and high terminals are available for optional bi- or triamplification (passive).

Now then, how does a pair of XRT24's sound? Basically neutral balance. Excellent dynamics and power handling. *Big* soundstage, which is what you'd expect with those tall line sources. Not very deep bass. Slightly coarse-textured highs, lacking the airy/silky quality of the most sophisticated tweeters (such as those in the Win SM-10, MACH 1 Acoustics DM-10, Snell Type A, etc.) and causing my Reservation No. 2.

I could be cruel and declare the basic design strategy of using 32 plain-vanilla aluminum-dome tweeters in a very expensive pair of speakers to be a mistake, but I don't want to be cruel because the engineering thinking behind the XRT24 is intellectually on a high plane as well as innovative in many respects, while free of tweako nonsense. To some well-heeled audiophiles the virtues of the speaker will be more important than its two shortcomings, especially since its relatively small footprint and high-quality furniture finish make it one of the few very tall speakers without the probability of over-my-dead-body spousal confrontations. I resolved my mental conflicts about it by moving it out of my main listening room/laboratory, placing it into my home theater system in the family room, and extending it on the bottom with the **Hsu Research HRSW12V** powered subwoofer. There, in combination with Mcintosh center-channel and surround speakers, the XRT24 is giving me total satisfaction. Is it because I'm less demanding of Bela Lugosi than of Béla Bartók?

Home Theater Loudspeakers

Mcintosh HT-3 and HT-4

Mcintosh Laboratory, Inc., 2 Chambers Street, Binghamton, NY 13903-2699. Voice: (607) 723-3512. Fax: (607) 724-0549. HT-3 THX 2-way, antiphase dipole, mirror-image, left/right surround loudspeakers, $2000.00 the pair. HT-4 THX 2-way center-channel loudspeaker system, $900.00 each. Review samples on loan from manufacturer.

These are the speakers referred to above, supplementing the XRT24 in my home theater system. They were sent to me several months later than the XRT24 and have not been measured yet as I am writing this. I can vouch, however, that their construction is of high quality and that they do their job very proficiently. The HT-4 center-channel speaker has three 1-inch aluminum-dome tweeters arrayed in a vertical line, which is pretty much a standard THX formation; not surprisingly this mini line source has a sonic signature very similar to that of the big line source in the XRT24, subtle but discernible. The HT-3 surround speakers appear to have no signature at typical surround-signal levels. My tentative impression is that the HT-3 and HT-4 are somewhat overpriced, but I am reserving final judgment until all the usual tests are completed. •

Loudspeaker Systems, from Good Value to Ultimate (and In Between)

By Peter Aczel
Editor and Publisher

Is the loudspeaker still the weakest link in the audio chain? Yes, undoubtedly, but the evidence of these tests is that the weak link is getting stronger all the time, in all price brackets.

Here's something to think about before you sink your teeth into the reviews. Loudspeakers are all imperfect to a greater or lesser degree; the imperfections are all different, so all the speakers sound at least somewhat different; therefore loudspeaker evaluation leaves at least some room for purely subjective opinion and speculation, much more so than any discussion of the incredibly perfect analog and digital signal paths of modern audio electronics. You'd expect most of the subjectivistic voodoo and tweako cultism in high-end audio to be about speakers—but no! Other than the wire/cable nonsense, which arguably relates to electronic events before the speaker terminals, most speaker discussions in tweako circles are relatively sober, certainly not comparable on the B.S. scale to, say, single-ended triodes. The inevitable conclusion: audiophiles retain the faculty of clear thinking only as long as the equipment is imperfect and there are real problems to be solved; when there aren't any, they panic and create imaginary problems.

Let's look at some solutions, then, from the real world of loudspeaker engineering.

Atlantic Technology System 250

Atlantic Technology International, 343 Vanderbilt Avenue, Norwood, MA 02062. Voice: (617) 762-6300. Fax: (617) 762-6868. System 250 six-piece home-theater speaker system, $1446.00 (plus minor options, stands, etc.). Tested samples on loan from manufacturer.

Home theater has somewhat different requirements when it comes to speakers than two-channel stereo for music. I am of the school that favors the most accurate stereo speakers for front left and right—never mind the "specialized" home-theater speakers that don't sound so great on music—but for the center and surround channels the rules are different (at least until AC-3 with its five full-range discrete channels, or something simlar, becomes the standard). The center and surround speakers are processors to some degree; they do not necessarily have to be "accurate" in the linear input/output sense. The home-theater subwoofer, on the other hand, must meet pretty much the same requirements as a stereo-system subwoofer. The Atlantic Technology System 250 is a remarkably cost-effective solution to the problem of excellent home-theater surround-sound performance without significant musical compromises. I never expected anything nearly as good for $1500.

Let's start with the costliest part of the system, the **252 PBM** Powered Bass Module ($569.00). It incorporates a 12-inch, paper-cone, long-throw woofer; an amplifier that offers the choice of three 40-watt (4Ω) channels, two of them for satellites, or 90 watts of power into the woofer alone; lowpass filter options of 80 Hz or 120 Hz; and various inputs and controls. It's quite nicely built, not too big—approximately 2 cubic feet in internal volume—and uses a vented-box design with two ports. The system is tuned to approximately 37 Hz; there is solid output down to 30 Hz, and even at 20 Hz the two ports put out some energy. Corner placement can further support the bass, so that in most rooms quite realistic theater-level low-frequency pressures are obtainable. I was impressed. Distortion certainly isn't extra low—this is not a motional-feedback design a la Velodyne—but the combined nonlinearities of the amplifier and the woofer are well within conventional limits.

The front left and right speakers, Model **251 LR** ($299.00 the pair), are of the so-called D'Appolito configuration: two 4-inch polypropylene midrange/woofers, one on top, one on the bottom, with a small plastic-dome tweeter in the middle. The sealed box is very small, about one third of a cubic foot internally; the optional metal stand tilts it slightly upward at seated listening height. Nominal impedance is 8 ohms. I found the frequency response of the 251 LR unusually interesting: up to 2 kHz it is dead flat, and I mean ±0.5 dB; from 8 to 18 kHz it is almost equally flat, perhaps ±1.5 dB; but between 2 and 8 kHz there is an almost perfectly symmetrical hollow, about 8 dB deep at its bottom! I have never seen anything like it and can virtually guarantee that it is not a design error but entirely deliberate. Somebody at Atlantic Technology doesn't like that "presence" range, that's for sure. (Maybe it's Peter Tribeman, their president—remember him from the old NAD days? He is known to have very strong opinions on sound.) I must say that the speaker sounds very sweet and creamy-smooth on all program material; maybe the frequency-response tailoring is in anticipation of harshness on most cinematic sound tracks. Even so, I prefer to have my equalizer elsewhere in the audio chain, just in case something is beautifully recorded in the first place. I find it remarkable that other reviewers of this 1994-vintage product never really got a handle on this fascinating peculiarity.

The center-channel unit, Model **253** C ($279.00), uses exactly the same drivers in a box of very similar size, except that the speaker is horizontally deployed on top of the TV and rests on a very simple but ingeniously designed base that permits about a 25° range of up-and-down tilting. In addition, there are two controls in the back marked "midrange timbre" and "hi-freq. level," which boost/cut the midrange/woofers and the tweeter, respectively, to permit matching the 253 C to front left and right speakers other than the 251 LR. A dot marked "251" shows the position of each control recommended for matching to the 251 LR; another dot marked "cinema eq." is the position supposed to "compensate for the natural brightness of many movie soundtracks," although the 251 position already suppresses the same two "brightness" octaves as the 251 LR. Thus doth the addition of two cheap passive controls an almost double-priced "Timbre Adjusting Center Channel Speaker" make—but why not, when the total six-piece system sounds better than it has a right to for the money?

The equally small Model **254 SR** surround speakers ($299.00 the pair) each have one 4-inch polypropylene driver on top, identical to those in the other units, plus two 3.5-inch "mid-tweeters" below, angled to fire frontward and rearward. The 3.5-inchers are phased to produce the "decorrelated" output needed for ambience channels. There is no need for dead flat response—indeed, for any significant degree of input/output accuracy—in ambience speakers, so I did not bother to make MLS measurements on the 254 SR, but it is fairly obvious that it has no top-octave (10 to 20 kHz) output. It does the job, and that's it.

I already mentioned in my Mitsubishi TV review in Issue No. 22 that the Atlantic Technology System 250, driven by Marantz MA500 power amplifiers, produced better movie sound than you'll hear in your typical small neighborhood theater. To be more specific, the sound is quite complete from *T. Rex* footsteps to violin harmonics; nothing is missing either on the bottom or the top; the dynamic range leaves very little to be desired, even in a fairly large room; no harshness or edgy quality intrudes on any kind of program material; in fact, you could do a lot worse than to use the System 250 as your main transducer for music, quite independently of home theater. I played a few of the Delos 20-bit CDs that sport the Dolby Surround logo and was quite genuinely satisfied, barely aware that I was listening to a tourist-class multichannel system, until I took the same CDs to my main listening room and replayed them through incomparably costlier speakers. Yes, they sounded quite a bit better that way, but the System 250 was not humiliated by the comparison. It is good enough not to have to apologize to much higher-priced home-theater speaker systems.

Such a conclusion brings up a cynical thought. Does it take no more than adding a 30 Hz bottom end and suppressing the brightness range to make a very modest loudspeaker array sound like a high-end system? (I never pretended I don't have a dirty mind.)

Hsu Research HRSW12V

Hsu Research, 14946 Shoemaker Avenue, Unit L, Santa Fe Springs, CA 90670. Voice/Fax: (310) 404-3848. HRSW12V powered subwoofer, $850.00 each (factory-direct, including shipping/handling). Tested sample on loan from manufacturer.

My review of the Hsu Research HRSW10 subwoofer appeared in Issue No. 19. The HRSW12V is quite similar conceptually but considerably upgraded in a number of ways, including the considerably higher price. The larger driver (a 12-inch unit instead of a 10-incher) and the dedicated outboard amplifier are the most obvious differences. The basic configuration remains the same: cylindrical enclosure made of lightweight but extremely strong and inert paper tubing, downward-facing driver, long duct tuning the vent; however, everything is bigger and beefier than in the 10-inch predecessor (which has meanwhile been dropped from the line to be reintroduced in powered form).

The HRSW12V cannot be evaluated from quite the same perspective as the HRSW10. The latter was incredibly low in price for a world-class subwoofer, $375.00 each (without amplification, to be sure), so that its performance exceeded all expectations. The factory-direct price of the HRSW12V, on the other hand, is only 22% lower than the retail price of the Velodyne Servo F-1200R, for example, which is also a 12-inch powered model but with accelerometer, motional feedback, and remote control. That means the HRSW12V is out of the bargain basement and on the main floor, competitionwise. Our review sample did not get to us early enough to be wrung out properly in the laboratory before this was written (we seldom get the early samples but prefer to have the last word anyway), so I am restricted to listening impressions here, which are much the same—in spades!—as told in my detailed and enthusiastic review of the HRSW10. This is a serious contender in the subwoofer Olympics and deserves to be judged against the high-priced entries. I wanted that much to be said even before further testing.

Dr. Poh Ser Hsu, the designer (see the aforementioned review in Issue No. 19 for his background), claims that his subwoofers outperform all others at 20 to 30 Hz when the SPL is very high. I personally believe that low distortion and good damping at 80, 90, 95, maybe 100 dB are more important than whatever happens at 115 dB; more about that in the next issue. I also intend to report on the recommendation by Hsu Research that the HRSW12V be placed in the listening area rather than near the main speakers—another subject on which I don't want to pass quick judgment. Meanwhile the HRSW12V is woofing prodigiously in my home-theater system.

Mcintosh XRT24

Mcintosh Laboratory, Inc., 2 Chambers Street, Binghamton, NY 13903-2699. Voice: (607) 723-3512. Fax: (607) 724-0549. XRT24 floor-standing 3-way column loudspeaker system, $7500.00 the pair. Tested samples on loan from manufacturer.

The review of this quite unorthodox and very interesting speaker system appears elsewhere in this issue as part of the feature article on Mcintosh Laboratory.

Paradigm Eclipse/BP

In Canada: Paradigm Electronics Inc., 101 Hanlan Road, Woodbridge, ON L4L 3P5. Voice: (905) 850-2889. Fax: (905) 850-2960. In the U.S.: AudioStream, Division of Bavan Corporation, MPO Box 2410, Niagara Falls, NY 14302. Voice: (905) 632-0180. Fax: (905) 632-0183. Eclipse/BP floor-standing bipolar 2-way loudspeaker system, U.S. $1799.00 the pair. Tested samples on loan from U.S. distributor.

The Janus-faced, i.e., bipolar, loudspeaker is popular among Canadian speaker designers—and I don't mean they favor two-faced deceit, just a bidirectional wave launch. Bipolar speakers are like two forward-firing speakers placed back to back and driven inphase; a positive-going signal makes both faces push; a negative-going signal makes both faces pull (assuming no internal phase inversion). The Paradigm Eclipse/BP is a representative upper-medium-priced Canadian specimen of the breed. It uses an 8-inch polypropylene-cone woofer and a 1-inch aluminum-dome tweeter on *each* face; the two woofers exhaust into the same columnar enclosure with a ducted port. Woofer-to-tweeter crossover is at 1.5 kHz. The whole affair stands a little over four feet high and is quite handsome, one might even say expensive-looking, in black gloss with runaround black grille cloth (other finishes available).

Contrary to my wonted procedure, I shall discuss the sound of the speaker first because it raises basic issues that the measurements alone do not. The soundstage is *huge,* undoubtedly as a result of the delayed/reflected rearward radiation. Certain audiophiles will make the Eclipse/BP their choice in this price range just for that. I feel about soundstage enlargement as I do about silicone implants—up to a point the effect may be glamorous but beyond that it's unnatural. The Eclipse/BP does something other than what the producers of my favorite CDs had in mind, that's for sure. I must admit, however, that regardless of the soundstage issue the speaker always sounds sweet and agreeable—indeed, too sweet and agreeable, even when the music is not. That would also make it certain audiophiles' automatic choice. To my ears the Eclipse/BP always sounds like a speaker, a very likable speaker to be sure, but not like an accurate replica of the program source. It has a bit more sonic "signature" than I like.

Now then, why? The edgeless, bland quality is quickly explained by the frequency response: the incredibly smooth tweeter, which stays within a ±1 dB strip from 2 kHz to 15 kHz, is set approximately 4 dB below the level of the woofer. I determined that with a 1-meter MLS measurement on the forward-firing axis; it is possible that the designers had in mind some kind of summed power response forward and rearward, but such things generally don't work out very precisely, and certainly not in my big room with its deadened rear wall. Interestingly, the axial response from 15 kHz to 20 kHz rises right back to the woofer level, obviously a "tailored" profile but of greater concern to my dogs than to me. The phase response is very well-behaved.

On the bottom end the vented system appears to be tuned, not very sharply, to approximately 23 Hz; the response is esentially flat down to 28 Hz or so. With four 8-inchers pumping, a stereo pair produces some pretty authoritative bass, making subwoofers superfluous except to incurable bass addicts like me. The impedance curve of the Eclipse/BP fluctuates between 3.7 and 15 ohms in magnitude, and between -20° and +35° in phase, presenting a relatively unchallenging load to the amplifier. Distortion at all levels is typical of nonexotic 8-inch woofers and 1-inch dome tweeters; nothing remarkable there. The crossover network drives the woofer and tweeter inphase, but a square pulse is not reproduced coherently at any acoustical summing junction. (No big deal; I've given up on the audibility of that.)

Wanting to isolate what I perceived to be that elusive signature of the Eclipse/BP, I excited both woofer and tweeter with tone bursts of all frequencies. Everything was squeaky-clean; no ringing, not even a trace. Tapping the cabinet with a padded mallet produced fairly dead sounds off the front and rear but more of a drumlike whomp off the larger side panels, even though high-tech bracing is claimed in the literature. Maybe that's what I heard as a signature, or maybe a combination of that and the tailored response and the bipolar launch. Maybe I'm strictly a monopole man. Maybe you'll love it. I certainly won't argue with you because a lot of very good engineering went into this speaker. It makes a statement.

Snell Acoustics Type A

Snell Acoustics, Inc., 143 Essex Street, Haverhill, MA 01832. Voice: (508) 373-6114. Fax: (508) 373-6172. Type A Music Reference System, $18,9999.00. Tested samples on loan from manufacturer.

What I wrote in my preview of the Snell Type A in Issue No. 22 still stands: this is the best loudspeaker system, overall, that I have tested and lived with. Please note that I said "overall," which is not the same as "in every way." For example, the woofer of the Type A is very good but not my favorite. From 80 Hz on up, however, I don't know of another speaker that satifies both my listening and technical requirements quite as completely.

The system—and I mean the stereo system, not the multichannel cinema version (see below)—comes in 7 pieces: two 63-inch high towers, 9 by 12 inches in cross section; two outboard passive crossovers, each as big as a bookshelf speaker; two refrigerator-sized subwoofers; and one cute little 80-Hz electronic crossover (made for Snell by Audio Alchemy). The latter is a very simple, stripped-down unit, called EC-200; the recommended electronic crossover for those who want everything to be "high end" is the Bryston l0B-PRO/Snell, a dedicated version of Bryston's $1195.00 super crossover. I tested both the EC-200 and the Bryston/Snell. (See Issue No. 16 for an early review of the basic Bryston model.)

Let's get the subwoofers out of the way first because, well, they're just not as interesting as the rest of the system. They are *big*, though: each enclosure measures 45 by 21.5 by 17 inches and houses an 18-inch driver loaded with a huge ducted port. The mouth of the port is 5 by 5 inches and the duct is *long*. Unfortunately, the center of the woofer is approximately 26 inches above the center of the port, making it extremely difficult to find a nearfield summing junction to measure the combined response of the two. I did the best I could and found essentially flat small-signal response down to the subsonic region, with the -3 dB frequency (f_3) at approximately 12 Hz. The box appeared to be sharply tuned to 15 Hz. Can't ask for much better.

The distortion measurements, taken at the flattest nearfield summing junction, were less impressive. With a 1-meter SPL of 90 dB, THD plus noise fluctuated between 0.5% and 5% from 50 Hz down to 10 Hz. The 0.5% minimum was at 20 Hz, the 5% maximum at 40 Hz. With the 1-meter SPL raised to 95 dB, the THD-plus-noise range was 0.3% to 5%, but this time the minimum was at 34 Hz and the maximum at 12 Hz. I would estimate the typical distortion at a random low frequency to be 2% at 90 dB and 1% at 95 dB, but that includes the inescapable residual room noise (probably the reason for the higher reading at the lower level). The motional-feedback Velodyne Servo F-1500R is at least 6 dB better at *any* frequency at the same SPLs and a whole order of magnitude (20 dB) better at *most* frequencies. (Other things being equal, the closed-loop system *has* to be cleaner than the open-loop system—the laws of physics say so.) The Velodyne is also somewhat better damped than the Snell; the latter exhibits just a bit of hangover when excited with low-frequency tone bursts (again, not unexpectedly in view of the fourth-order tuning of the vented box).

Please don't misunderstand me; the Snell subwoofer is no slouch; it goes all the way down to the *T. Rex* frequencies, can play really loud, and has no more distortion than is reasonable in an open-loop design. Don't refuse it as a birthday present. It's just that the Velodyne represents a newer and more sophisticated generation of subwoofer design, in a mere 20-inch cube and at a lower price, with amplification thrown in. (A single unamplified Snell SUB 1800, if purchased separately, costs $2499.00; that's $904.00 more than the amplified Velodyne.) I removed the subs from the Type A system, substituted a pair of Velodyne Servo F-1500R's, and after some crossover fussing (more about that in a moment) was a thoroughly happy camper. Will Snell sell you a subwooferless Type A Music Reference System for $14,001 (= 18,999 - 2 x 2499)? I see no reason why not, as long as you don't involve them in the substitution.

About the electronic crossovers—decisions, decisions. I found the EC-200 and Bryston l0B-PRO/Snell to be equally low in distortion (meeting the spec of 0.005%, i.e. -86 dB, at 1.5 V output, 20 kHz highpass and 20 Hz lowpass) and equally accurate in lowpass and highpass filter contours. The Bryston is better built and a little more convenient to set for level; the EC-200 once gave me a frighteningly loud pop as I was turning off various parts of the system, but I was unable to duplicate the same circumstances again. (I suspected dc offset, but the measured amount was minimal. Weird.) Is it worth the additional expenditure to get the Bryston? Some people will undoubtedly want it. (No, there is no difference in sound—you ought to know better than to ask such a question.) The electronic crossover frequency is set at 80 Hz; the highpass filter is second-order (12 dB per octave), the lowpass filter fourth-order (24 dB per octave). With the Velodynes I left the lowpass section of the electronic crossover unconnected, using the lowpass and volume controls on the back of each subwoofer instead; it's not an exact match to the Snell lowpass slope but, after a bit of fussing and measuring, "close enough for government work" as the saying goes. Crossovers at 80 Hz are not sensitive to exquisitely subtle adjustments.

You've been waiting for the main course all this time, and with the mundane hors d'oeuvres and soup out of the way I'm ready to serve it to you: yes, the towers! In my not-so-humble opinion they're a truly superior piece of engineering. Each tower houses four 6.5-inch drivers (call them upper woofers or lower midranges), two 5-inch midrange drivers, a single 1-inch textile-dome tweeter, and a single rearward-firing 1-inch metal-dome tweeter. The seven forward-firing drivers are deployed symmetrically in a vertical array: tweeter in the center, one 5-inch midrange above and one below the tweeter, two 6.5-inch drivers on top of the array and two on the bottom. The huge outboard passive crossover directs the signal traffic to the drivers via a massive 8-cable umbilical cord; the crossover frequencies are 350 Hz and 2.8 kHz. Four very precise tweeter-level settings are provided (+1 dB, flat, -1 dB, -2 dB); the rear tweeter can be switched on or off and trimmed for level (by ear) with a continuous control.

The amazing thing is that this complex structure acts as a monolithic supersmooth transducer, with seamless and almost amplifier-flat response from 80 Hz to 20 kHz. My measurements (MLS from 300 Hz up, taken at 2 meters as well as 3 meters; quasi-nearfield from 80 Hz to 300 Hz; everything set flat) corresponded very closely to the manufacturer's spec of ±1.5 dB over that entire range. In the crucial range from 1.5 kHz to 11 kHz, the response is better than ±1 dB—and that with the grille in place! (Unfortunately, a snap-on grille frame proved to be too much of a manufacturing problem, so the grille cloth is unremovable.) At 30° off axis, all that happens is that the forward-firing tweeter rolls off above 11 kHz with a slope of about 12 dB per octave. Below 11 kHz there's hardly any difference between on-axis and off-axis response. Some speaker.

And that's not all. The distortion generated by the towers—as distinct from the Snell subwoofers—is almost Velodyne-low. At a 1-meter SPL of 90 dB, as referenced to 150 Hz, the nearfield THD + N of the upper 6.5-inch driver hovered between 0.1% and 0.2% from 300 Hz down to 125 Hz, rising to 1.5% at 80 Hz. The upper 5-inch driver, measured the same way, with the 1-meter SPL at 90 dB as referenced to 1 kHz, yielded 0.2% to 0.5% in its range of 350 Hz to 2.8 kHz, averaging about 0.3%. The tweeter, again at 90 dB SPL at 1 meter as referenced to 5 kHz, remained near 0.3% over most of its range, dropping to 0.2% above 6 kHz. I didn't even bother to take my usual 95 or 96 dB measurements because, as already mentioned above, the distortion almost invariably reads lower at the higher SPL on account of the constant room noise. These figures nudge the performance of the "distortion-free" Velodyne DF-661 without quite equaling it—but the Snell is flatter (indeed, flatter than just about anything).

As for ringing or storage—nothing. All the tone bursts I tried looked squeaky-clean. (That textile-dome tweeter is especially nice—and especially interesting coming after all the Snell models that use metal-dome tweeters.) The impedance curve of the tower shows rather wide swings in both magnitude (3.3 ohms to 20 ohms) and phase (-55° to +30°), not surprising in view of the complex crossover. A high-quality (nontweako) solid-state amplifier should have no trouble driving that kind of load. Incidentally, the terminal configuration on the outboard crossover also permits biamping or—if you'll pardon the expression—biwiring. (For the benefit of new readers: biwiring is hogwash but harmless.)

Another interesting aspect of the design is that neither the towers nor even the SUB 1800's are so heavy that you need piano movers to experiment with speaker placement. Designer Kevin Voecks believes that 1.5-inch walls in a speaker box yield no significant sonic benefits but make life miserable for the audiophile. (Of course, that's what certain techie masochists want.)

All right, all right, I'll tell you about the sound. The speaker has virtually no sound of its own—that, of course, is what makes it great. It adds no signature to the program material, so the latter's own signature (hall acoustics, microphone characteristics, recording setup, etc.) emerges with unprecedented clarity. The wave launch is medium-sized, and the soundstage is not gigantic on typical program material—but is "Wow, what a huge soundstage!" the right impression, or even a desirable impression? The Type A tower sounds more like an idealized 14- or 15-inch coaxial speaker, say like the old Win SM-10, only bigger—and even that is an inadequate description. There is certainly soundstage depth and width but not of the in-your-face, check-it-out-dude kind. What's more, the sound is never harsh, from *ppp* to *fff*, even though the high frequencies are certainly not rolled off. I guess what I'm trying to say, without resorting to the tweako reviewers' quasi-pornographic vocabulary, is that the speaker sounds beautiful—more like the real thing than like a speaker. (It's easier to be highly articulate and specific about something bad than something this good.)

I first met Kevin Voecks in 1977, when he was barely out of his teens. (Maybe not even.) At that time he was a classic audio tweak, fondly indulgent of his own subjectivity. Now, 18 years later, he is one of the most scientific, practical, self-critical, and verifiably effective speaker designers, with more than a few world-class designs to his name. To say that I am impressed would be a serious understatement. I only hope that he will soon do an "A Minor," incorporating the most important aspects of the Type A design in a more compactly packaged and

(continued on page 38)

Catching Up on Analog and Digital Electronics for Stereo

By Peter Aczel
Editor and Publisher

Two-channel stereo is alive and well and living in the living rooms and family rooms of music lovers. Not all sales in the audio world are home theater, at least not yet.

Some of the reviews below should have appeared in the last issue but were held over; others are of more recently received equipment.

Outboard D/A Converter with HDCD
EAD (Enlightened Audio Designs) DSP-7000 Series III

Enlightened Audio Designs Corp., 300 West Lowe, Fairfield, 1A 52556. Voice: (515) 472-4312. Fax: (515) 472-3566. DSP-7000 Series 1I1 outboard D/A converter, $2495.00 (optional balanced outputs $399.00 extra). Tested sample on loan from manufacturer.

This is basically the same processor as the DSP-7000 Series II reviewed (and recommended with minor reservations) in Issue No. 20, with one important change. EAD has made a questionable marketing move and incorporated the Pacific Microsonics HDCD decoder chip in all of their latest digital processors, including this one. I say questionable because the HDCD (High Definition Compatible Digital) encode/decode process has so far been endorsed only by the tweako/mystico element in the high-end community and is regarded with considerable suspicion by the best brains in the business. No major label is currently releasing HDCD-encoded CDs, nor will they as far as I know (except perhaps in isolated cases where a recording artist absolutely insists on it). Reference Recordings is the showcase label for the system, and the hardware customers for the decoder chip are the familiar roster of boutique manufacturers catering to the high-end ghetto.

One reason for the suspicion of the best-educated practitioners is that Pacific Microsonics has never made a full disclosure of the exact operation of the system, from the input of the encoder to the output of the decoder. That's not the way Dolby became a standard. Yes, there's a patent, but as this is being written no one has seen it in print yet; meanwhile the Pacific Microsonics people have become experts at saying a lot and revealing nothing. The only really good source of information on HDCD so far has been the European patent application, which is not a privileged document over there and has been read by a number of authorities whom I trust. What I hear from them is that HDCD involves all sorts of unnecessary and unjustified overprocessing that, in their opinion, no "enlightened audio designer" would want to subject the music to.

This is not the time and the place for a critique of HDCD, which will have to wait until the most detailed information is available and the subject is crystal clear; maybe then a more favorable picture will emerge. The above remarks are simply a reflection of my present misgivings. I don't believe that 16-bit linear PCM needs "fixing," especially when the new 20-bit masters make it even easier to end up with all 16 bits on the CD.

To get back to the DSP-7000 unit itself, the discernible changes from Series II to Series III are these:

• The HDCD decoder chip (PMD100) has replaced the NPC SM5813 digital filter, allegedly improving the playback of non-HDCD-encoded discs as well.

• On the front panel a blue light turns on when HDCD decoding is automatically activated by an encoded disc. (A very old and slightly nasty ethnic joke from New York pops into my strictly non-PC and non-HDCD-impressed mind: "Abie, turn on the blue light. The customer wants a blue suit.")

• The reclocking circuitry has been upgraded following David Rich's free advice in Issue No. 20, p. 50: it is now a phase-locked loop, utilizing a voltage-controlled crystal oscillator. (Gee, why don't the EAD people use

Robert Harley as their digital consultant?)

• The analog output level in response to a full-scale (0 dB) digital input without encoding has been reduced by 6 dB to 1 V. Apparently this is a nonnegotiable requirement for HDCD licensing by Pacific Microsonics. They must be afraid that non-HDCD playback would sound *better* than HDCD because it is *louder.* (The official justification is of course the convenience of equal levels from all sources, but audiophiles know that slight volume-control adjustments are always needed, regardless, depending on how "hot" the recording is—so what's wrong with trimming the volume manually for HDCD?) I find this feature particularly obnoxious because it reduces the 16-bit system to 15 bits and skews the full-scale THD + N versus frequency measurements; luckily, if you have an Allen wrench of the right size to open up the unit, a single DIP-switch setting on the circuit board will defeat the modification and restore the normal 2 V output. That's what I did (with the EAD chief engineer's vigorous consent, believe it or not), and that's how I made my measurements.

Lo and behold, the performance of Series III with our usual digital test signals was not as good as that of the pre-HDCD Series II. Now, it is an outside possibility that our sample had slightly substandard DACs, although Burr-Brown's quality control of their flagship PCM63P-K is known to be virtually flawless. Perhaps EAD had tweaked around with some circuit adjustments, but I doubt it. In a few other published test reports the Pacific Microsonics PMD100 chip shows no tendency to degrade the non-HDCD performance. Bottom line: I have no explanation, but here's what I found.

Full-scale THD + N measured -92 dB to -91 dB at the lower frequencies (i.e., 6 dB to 7 dB in excess of the theoretical ideal of -98.08 dB), dropping above 1 kHz to a minimum of -94.4 dB at 4 kHz and then rising again. Series II had been approximately 3 dB better below 1 kHz. With a -20 dB digital input, Series III still showed 1 dB to 2 dB excess distortion, whereas Series II had shown virtually none (maybe 0.5 dB). Most dramatic was the difference in gain-linearity error: Series III was off by +2.3 dB at -100 dB and by +0.7 dB at -80 dB in its less good channel; the "better" channel measured +1.25 dB at -100 dB and +0.3 dB at -80 dB. Series II had displayed virtual perfection on this test, with no more than 0.1 dB error down to -100 dB. Here comes the mindblower. Gain linearity below -70 dB is *always* better with dither; indeed, it shouldn't even be measured without dither—but Series III is the exception! Down to -90 dB it was just a hair better without dither! Figure that one out. I couldn't; I kept muttering something about the dither in the PMD100, but what do I know? I also did what I now call the Rob Watts test, an FFT plot of a dithered 1 kHz tone at -60 dB (see Issue No. 22, p. 36), which should show a super clean -130 dB noise floor without any harmonics. Series III showed 2nd and 3rd harmonics 18 dB and 20 dB above the noise floor, and there were lots of higher harmonics as well. Not as clean as the best. The noise spectrum with digital zero input was also 5 dB to 7 dB worse at many frequencies than that of Series II, although it was still very good. One thing that hasn't changed from Series II to Series III is the radiofrequency interference (RFI). Even in the standby mode, just plugged in but not operating, the unit is absolute death on FM reception. Leave it unplugged when not in use, or have your FM tuner several rooms away from it.

I listened to the Reference Recordings *HDCD Sampler Volume 2* demo CD to see if I would come under the spell of decoded HDCD sound, but my impressions were inconclusive. Keith O. Johnson makes good recordings, as we all know, so most of this stuff sounds pretty nice. I think he has been leaning toward a "wetter" sound lately than I consider ideal. In any event, the hall and the microphone setup have a much greater influence on the sonic end result than the use or nonuse of HDCD. The three pairs of tracks demonstrating use versus nonuse merely prove that there is a difference. (If there weren't a difference, after all the hype, it would be the ultimate tweako exercise in unreality, but that's not the case.) But is the HDCD encoded/decoded sound clearly better? To give a definitive answer, I would have to go to a live recording session where there's a switchabe HDCD encode/decode loop in the direct microphone feed. I have no way of knowing exactly what RR did to those tracks. HDCD seems to soften the attacks and somehow defocus or airbrush or pastel-color the details, as it were—"better" sound to some, a "processed" sound to others, and vague perceptions in any event. Controlled listening tests are in order, but they must be part of a total, in-depth evaluation of HDCD.

Stereo Power Amplifier
Mcintosh MC500

Mcintosh Laboratory, Inc., 2 Chambers Street, Binghamton, NY 13903-2699. Voice: (607) 723-3512. Fax: (607) 724-0549. MC500 stereo power amplifier, 500 watts per channel, with meters, $6500.00. Tested sample on loan from manufacturer.

The review of this no-holds-barred super amplifier appears elsewhere in this issue as part of the feature article on Mcintosh Laboratory.

Mono Power Amplifier
Sentec PA9

Sentec America, Route 9, Stockport, NY 12172-0329. Voice/Fax: (518) 828-8490. PA9 class AB monaural power amplifier, $600.00 each. Tested samples on loan from manufacturer.

Made in Sweden and reflecting the Scandinavian taste for simplicity, moderation, and quality in all aspects of design, this small mono amplifier costs twice as much as the Marantz MA500 (see Issue No. 22, pp. 27-28), has half as much power output before clipping, about the same distortion at low power but much higher distortion on moderate peaks and beyond; it is, however, small and cute (8 in. by 3 in. by 10 in. deep) and much better constructed. Its slightly exotic appearance and provenance will make some audiophiles believe it "sounds better."

David Rich examined the circuit schematic and sent me a memo that I might as well quote:

"Your basic fully complementary amplifier. Complementary degenerated differential pairs are driven by single-transistor current sources. The current sources have LED voltage references. The differential pairs are resistively loaded. The second stage is also a degenerated differential pair. Each side of the second differential pair inputs is connected to the first differential pair outputs. The use of two sets of differential pairs significantly improves the common-mode rejection ratio (CMRR) and hence the distortion that results from common-mode signals (see the Harman Kardon PA2400 review in Issue No. 21). A single transistor is used to bias the output stage. The unit has no predriver stage. The output stage has three paralleled devices. The current limiter is the standard one-transistor design. No C_1, but C_2 is present (see Issue No. 20, p. 28, Fig. 6). The mono channel uses 21 transistors and 6 diodes."

The THD + N versus watts measurements were quite unusual in that the distortion isn't noise-dominated. Probably because of the lack of a predriver stage, there is significant nonlinearity rising with level, long before clipping. Minimum distortion is reached in the neighborhood of 1 watt, where it is -90 dB with an 8-ohm load and -86 dB with a 4-ohm load. Clipping occurs at 72 watts into 8 ohms, with -68 dB distortion before the breakpoint, and at 120 watts into 4 ohms, with -63 dB distortion. These numbers apply to 20 Hz and/or 1 kHz; the 20 kHz measurement is worse by 7 dB to 10 dB at all levels except the lowest—dynamic distortion at its most obvious.

The PowerCube test (see Issue No. 20, pp. 16-17) drew a fairly decent picture at 8Ω/4Ω/2Ω magnitude and -60°/-30°/0°/+30°/+60° phase, with the exception of 2Ω at -60°/+60°, where the beginning of serious current limiting was visible. The current limiting becomes severe with a 1Ω load regardless of phase, except 0°. With the one-transistor current limiter that was predictable. Maximum dynamic output voltage was just a little over 27 V into the five 8Ω test loads, declining with a linear slope to 18 V into 1Ω/0° and much more severely into the reactive loads. The calculated dynamic headroom at 8Ω is 0.65 dB. Peak current output measured 13 amps. Thus the overall PowerCube performance is unimpressive but still better than that of the average small amplifier.

I see little reason to favor the Sentec PA9 over the Marantz MA500, except for the PA9's quality/size ratio.

Line-Level Stereo Preamplifier

Sentec SC9

Sentec America, Route 9, Stockport, NY 12172-0329. Voice/Fax: (518) 828-8490. SC9 line-level stereo control amplifier, $700.00. Tested sample on loan from manufacturer.

This is the preamp sold by Sentec to complement a pair of PA9's. It is also small and cute, about half the size of a normal preamp and rather nicely finished. It has six line-level inputs, two tape monitor loops, a headphone jack, and separate balance and level controls. The the two tape monitor loops are switched in such a way that you can't put a tape recorder into a self-connected loop and cause destructive oscillations—a quality feature. If you wish, you can listen to one source and record another. A relay with time delay at the output prevents turn-on thumps.

The circuitry is discrete and somewhat complex, with many similarities to the PA9. David Rich examined the schematic and summarized the design as follows: "JFET input with current source used for the tail and resistive load. The output goes differentially to a bipolar differential pair with degeneration. The load is an active bipolar current mirror. A light-emitting diode (LED) is placed in series with one side of the differential pair's collector and the current mirror output. This provides the bias voltage for a class A bipolar emitter-follower output. Direct-coupled; dc offset is controlled with a pot. Medium feedback and good design techniques; it should have low distortion." (It does; see below.)

The design features David didn't like he noted thusly: "A single power supply for both channels. Low rail voltage, ±10 V. Zener diodes are placed across the regulated rails. The unregulated rails go to this point through a resistor. No active regulator circuits. (I have seen the same thing in $200 Pioneer receivers, where cost cutting made a lot of sense; why in a $700 preamp?) Expect proneness to line hum. On the positive side, the preamp has RFT filtering in series with the line; I think Sweden has tough RFI suppression requirements, now coming to all of Europe."

Now David may have his faults but being wrong isn't one of them. Indeed, before I found a way to ground the unit optimally in my test hookup, the sensitivity to input lead dress was astonishing—moving the input cables a couple of inches changed the THD + N by a factor of 6, or even more! That was all N, namely hum. After I had that gremlin figured out and under control, the results were truly excellent: -95 dB to -97 dB distortion at all frequencies between 1 V and 2 V output; slight dynamic distortion discernible at 20 kHz beyond 2 V; clipping at just over 5 V. Even at 20 mV out, which is 40 dB below the 2 V line-level reference, the THD + N at any frequency was -61 dB or lower. Changing to a 600-ohm load had negligible effect. Can't ask for much better.

Crosstalk was measured with 1 V in, the level control set for unity gain (1 V out), and the balance control at its detent. The results were excellent; right leakage into left was 5 dB to 14 dB lower than left into right, but even the latter was as low as -68 dB at 20 kHz, falling to -100 dB at 200 Hz and below. The better channel went as low as -105 dB at the lower frequencies and did not rise higher than -82 dB even at 20 kHz. These numbers are right up there with the best.

One must conclude that the minor circuit compromises in the Sentec SC9 are academic because they have little or no effect on the actual performance, which is comparable to that of just about any high-end preamplifier. Highly recommended.

Outboard Phono Preamplifier

Sentec PP9 RIAA

Sentec America, Route 9, Stockport, NY 12172-0329. Voice/Fax: (518) 828-8490. PP9 RIAA phono preamplifier, with MM9 moving-magnet input board, $400.00; MC9 moving-coil add-on board, $100.00. Tested sample on loan from manufacturer.

To expand the SC9 for phono playback, you plug this little brick-shaped black box, only slightly larger than the average modem, into one of the line-level inputs. An external transformer that plugs into the wall powers the unit. There is no power switch—it's the just-leave-it-on-it-hardly-draws-any-current philosophy—and no relay to prevent turn-on thumps should there be a power interruption/resumption.

David Rich's notes on the circuit: "A two-stage design. Topologies are the same as in the SC9, except that the bipolar follower stage is missing . Direct-coupled with dc servo in second stage. Better power supply, with ±11 V rails and an open-loop bipolar follower after the zener diode. Uses the floating ground arrangement also used in the Sentec DiAna outboard DAC. (The power transformer has no center tap.) RFI line filter and surge protection on secondary side of transformer."

The bench measurement results were a mixture of excellent and merely good. The RIAA equalization was truly excellent: ±0.075 dB from 30 Hz to 20 kHz; at 20 Hz it was off by -0.2 dB. Crosstalk was also excellent; with a 5 mV input (RIAA pre-emphasized), it fluctuated between -65 dB and -96 dB. No phono cartridge comes even close to that in channel separation. Input-referred noise measured 0.6 µV, and the spectrum of the noise floor as measured at the output hovered around the 5 µV level at most frequencies. I would call that very good but not great. THD + N at all but the lowest frequencies was -80 dB just before the onset of clipping, which occurred at 6 V. The 20 Hz measurement tracked with the higher frequencies up to 2 V out, after which it worsened by a maximum of 10 dB—purely academic. I have seen some -90 dB phono preamps, but this is pretty good THD + N on the whole.

All of the above results apply to the MM9 board; the MC9 board was not supplied, but David Rich had this to say about the MC9 schematic: "The moving-coil amplification stage is a 4-transistor open-loop affair. Because it has no feedback it uses a complementary topology that gives a 3 dB noise penalty."

Bottom line: if you go with the Sentec SC9 and need phono, by all means get the PP9 as well.

Compact Disc Player

Sony CDP-XA7ES

Sony Electronics, Inc., 1 Sony Drive, Park Ridge, NJ 07656. Voice: (201) 930-1000. Fax: (201) 930-4748. CDP-XA7ES compact disc player with RM-D921 remote control, $3000.00. Tested sample on loan from manufacturer.

This is now the top of the Sony ES line of CD players and a direct continuation of the design progression started with the CDP-X779ES (see Issue No. 18) and continued with the CDP-X707ES (see Issue No. 21). All three models are essentially the same electronically, so there is no need to explain the basic design all over again.

The outstanding feature of the circuit remains the CXD2562 pulse D/A converter, still unsurpassed and possibly even unequaled by any other in several areas of performance. In the XA7ES, the DAC is further improved by coupling it to a new chip, the CXA8042, whose ultimate function is to improve the PSRR and whose mode of operation allows the Sony marketing people to put "Current Pulse D/A Convert System" on the front panel of the XA7ES to distinguish it from the very similar X779ES and X707ES, which by the same argument have voltage-pulse D/A conversion. Pretty subtle difference.

The analog filter at the single-ended output is also a little different in the XA7ES; the change is from a generalized impedance converter (GIC) structure to a more standard Sallen-and-Key filter built around the output stage. The filter at the balanced output remains Sallen-and-Key, as before. (Is this the end of Sony's advocacy of the GIC solution?)

Much was made of the debut of the so-called Score digital filter IC in the X707ES, but the XA7ES uses yet another ostensibly improved filter chip, the CXD8504M, which is said to have a wider data path than its predecessor. What hasn't changed, on the other hand, is that the single-ended output still uses a more sophisticated circuit than the balanced output. Actually, very little has changed in terms of circuitry.

The mechanical and cosmetic changes are greater. The unit is now a very professional-looking black and has a keypad on the front panel, not just on the remote.

Most significantly, the laser tracking assembly has been changed to a fixed pickup design. The rigidly mounted optical block remains stationary as the rotating disc moves past it on a stable sled. A brass stabilizer, reminiscent of the phono era, holds the rotating disc in place, providing damping and precise centering. If you mislay the stabilizer—tough. The machine won't play. If you drop it, you can break or dent something with it, as it weighs well over a quarter of a pound. That's the price you pay (not to mention the $3000) for a mechanism copied directly from Sony's CD mastering equipment and claimed to increase the quality of the data stream picked up by the laser, with less dependence on interpolation or averaging. I couldn't measure any improvements over the already flawless transport of the X707ES but was duly impressed by the massive deluxeness of the whole machine.

Actually, I couldn't measure any improvements, period. That was no surprise because both the X779ES and the X707ES measured within a hairsbreadth of theoretical perfection, and so does the XA7ES. Sony has achieved the highest level of CD playback performance known to me. All other designs, CD players as well as outboard D/A converters, show some gain-related analog distortion as the digital signal level is raised from -30 dB or -20 dB to full scale (0 dB). Only the Sony units approach the theoretical 16-bit limit of -98.08 dB on full-scale THD + N. The excess distortion produced by the XA7ES at full scale is of the order of 1 to 2 dB across the audio spectrum, pretty much the same as in the case of the earlier Sony units but *no others* (not even the costliest outboard processors, which never do better than 4 dB and often do worse). At the -24 dB level the excess distortion is about 0.5 dB. The single-ended and balanced outputs measure about the same.

Other measurements are of comparable perfection. Quantization noise: -97.6 dB. Dynamic range: 97.5 dB. Channel separation: 115 dB at 16 kHz, increasing to 132 dB at low frequencies. De-emphasis error: unmeasurable (less than 0.02 dB). Low-level linearity error: 0 dB down to -100 dB and beyond. Spectrum of 1 kHz tone at -90 dB, with dither: no trace of harmonics. No wonder they have to reach to come up with a new top-of-the-line model almost every year.

In operation the XA7ES is smooth as silk, just like its predecessors; I like the extra convenience of the keypad on the front panel and grumble occasionally about the brass stabilizer when I can't find it instantly. Everything considered, this is CD-player heaven. Of course, less expensive players *sound* just as good, as our regular readers very well know, but some audiophiles will want what is provably best in every way, and this is it. Note, however, that the original X779ES cost only $1900; we're now up to $3000, a 58% increase. That's not just the yen exchange rate and not just the new features; that's high-end marketing.

Stereo Power Amplifier

Sunfire **(follow-up)**

Sunfire Corporation, P.O. Box 1589, Snohomish, WA 98291-1589. Voice: (206) 335-4748. Fax: (206) 335-4746. Sunfire "Load Invariant High Fidelity Stereo Power Amplifier," $2175.00. Tested sample on loan from manufacturer.

The preview of Bob Carver's truly different Sunfire amplifier in Issue No. 22 was based on a preproduction (or very early production) sample that we had our hands on for just one day. This follow-up review is the result of testing a completely current production unit. My description of the basic features of the Sunfire and David Rich's analysis of the design concept and circuitry are still entirely valid as published in the preview and need not be repeated here.

As the cliche goes, there's good news and there's bad news (no, this is not a joke). The good news is that the 43-pound Sunfire, at $2175, does basically the same thing as the 165-pound Krell KSA-300S at $9500. Both deliver *at least* 49 volts per channel continuously into *any* load. That means at least 300 watts into 8 ohms, 600 watts into 4 ohms, etc., doubling the power into half the load impedance. (It's possible that the Krell holds up a little better into 1 ohm or less, but by then who's counting?) That, essentially, is the Bob Carver advantage: no limit to cheap watts; no load too tough; no hernia. The bad news is that the Sunfire does all that at rather high distortion.

Now, it should be added that Bob Carver doesn't consider the amount of distortion produced by the Sunfire to be bad news at all. Ultralow distortion isn't one of his priorities, and there is good support for his position. The threshold of THD audibility on pure sine waves, under ideal test conditions, is believed to be in the neighborhood of 0.2% (-54 dB); on music it's much higher. Furthermore, above 10 kHz fundamentals all harmonics are inaudible, whatever their level. The Sunfire doesn't break the assumed thresholds, although it comes close. The position of *The Audio Critic* has always been that distortion is the measure of accuracy and should be kept as low as possible within the limits of budgetary sanity. When we have 16-bit clean digital equipment like the Sony CD player reviewed above, do amplifiers need to be only 9-bit clean in terms of distortion at full output? Amplifiers that are 16-bit clean just before clipping do exist. Bob Carver says that's nice but unimportant.

Having been told that a lot of the measurable THD + N in the output of the Sunfire was ultrasonic switching noise (i.e., inaudible N), I used a measurement bandwidth of 22 kHz instead of the usual 80 kHz for most of my bench tests. With a 22 kHz cutoff, my highest test frequency was 6.5 kHz, in order to include the second and third harmonic (13 kHz and 19.5 kHz) with some lowpass margin to spare. Later I found out that in my up-to-date review sample the switching noise had been reduced to a fairly low level and was well above 80 kHz in any event, so that the THD + N readings at high frequencies represented mainly dynamic distortion, not noise. I did a 0-to-80-kHz FFT analysis on a 19.5 kHz test signal at 200 watts into 8 ohms and saw 2nd/3rd/4th harmonics at -57/-53/-72 dB (plus three harmonically unrelated blips in the -85 to -93 range) but found the overall noise floor to be very low, centering on the -110 dB line. According to David Rich (who has familiarized himself with the Sunfire circuit and admires many of its features), slightly costlier input op-amps would have kept the distortion down, but as I said that wasn't one of the designer's priorities.

To get deeper into the numbers, the amplifier clips at 310 watts into 8 ohms and 620 watts into 4 ohms. That's true, with very small corrections, for all frequencies and either channel. The THD + N readings show a generally falling tendency with rising output, up to clipping, but not nearly as regularly and predictably as seen in conventional amplifiers with purely noise-dominated distortion. In the Sunfire curves there is evidence of level-related nonlinearity here and there. Worst-case distortion at high output: -47 dB at 20 kHz just before clipping, with the 80 kHz measurement bandwidth. But even with a 6.5 kHz input and the 22 kHz measurement bandwidth, the distortion is no better than -57 dB from 1 watt up to clipping in the less good channel, regardless of load impedance. At lower frequencies the picture improves; the 20 Hz and 1 kHz readings are 10 to 15 dB lower at high output with the 22 kHz cutoff, slightly varying with load impedance and not quite the same in both channels. Absolute best-case distortion using the 22 kHz cutoff: -81 dB at 50 watts into 8 ohms, at the low point of an irregular dip in the 20 Hz distortion curve of one channel; the other channel hits -80 dB just before clipping but not at 50 watts, and neither channel goes quite as low with a 4-ohm load. By comparison, using the 80 kHz cutoff, the lowest reading for 20 Hz or 1 kHz is -64 dB, just before clipping into 8 ohms. Compare these results with any of those reported in our power-amplifier survey in the last three issues and draw your own conclusions.

(It should be noted that the 22 kHz and 80 kHz filters referred to above have flat response in their passband; they should not be confused with the humpbacked A-weighting filter Bob Carver uses for some of his tests.)

The Sunfire's square-wave response shows normal rise times, flat tops, and a bit of overshoot in the corners.

In Issue No. 20 we introduced the PowerCube test; the reader is referred to pp. 16-17 of that issue for a complete explanation and illustrated examples. The Sunfire drew a virtually perfect PowerCube at 8Ω/4Ω/2Ω magnitude and -60°/-30°/0°/+30°/+60° phase. It delivered approximately 56 V of the standard test signal (1 kHz bursts of 20 ms duration) into those 15 test loads at the targeted 1% (-40 dB) distortion. At 1Ω magnitude and the same five phase points the voltage declined to values between 40 V and 29 V, either because 56 V (= 3136 W into 1Ω) would have yielded more than the targeted 1% distortion, or because the PowerCube hardware/software got nervous at that level (I've seen that happen before), or because of some other unidentified glitch. I don't quite trust the 1Ω readings and am willing, on a benefit-of-the-doubt basis, to declare the amplifier to be indeed "load invariant" as it says on the front panel. It is interesting to note that a comparison of the PowerCube numbers and the continuous power curves (at 1% distortion, not at the breakpoints) shows that the circuit design allows no dynamic headroom as such—and who needs it with this kind of continuous output capability?

Very new readers of *The Audio Critic* may now ask how the Sunfire sounds, but the regulars know better. The amplifier has dead flat response within the audio range, produces monstrous output levels without clipping, and doesn't distort above the threshold of audibility, so how could it have a sound? That applies, however, only to the "voltage source" output terminals (see Issue No. 22, p. 31); at the "current source" terminals there is a big 1-ohm resistor in series with the speaker load, and that definitely changes the sound, in exactly the same manner as a tube amplifier with high output impedance. The 1-ohm source impedance interacts with the loudspeaker impedance to skew the frequency response (see Issue No. 16, p. 55, Fig. 8, and the discussion on p. 56). You can do that yourself with any amplifier and a 1-ohm resistor, but Bob Carver gives it to you built in, with the option not to use it. Don't tell him, though, if you're not going to use it; a tag that hangs on every newly unpacked Sunfire amplifier says: "Stop! Stop! Stop! For best sound and largest soundstage, biwire woofer to *voltage source* (transistor amp output) and high-frequency drivers to *current source* (vacuum tube output)." Tweako dealers and their customers find the resulting softening of the highs and slight "warming up" of the mids/lows to be magical; to me it's just an uncontrollable tone control, sometimes for the better, sometimes for the worse, depending on the speaker and the program material. I favor the accurate and predictable voltage-source outputs because I can diddle with the frequency response in other ways if I so desire.

That's the long and the short of it. The Sunfire is a brilliantly designed, extremely powerful, highly cost-effective, quite idiosyncratic, and decidedly controversial amplifier. It reflects Bob Carver's priorities, which are not everybody's. It should really be compared with the Carver Corporation's higher-priced Lightstar amplifier, which was started by Bob before he left and finished by others. Unfortunately, our repeatedly promised review sample never arrived; meanwhile the Carverless Carver Corporation (Hamlet without the Prince of Denmark, in Sir Walter Scott's immortal quip) isn't exactly setting the audio world on fire. I can wait. •

Twice Shy: On Reencountering Multichannel Music Formats

By Daniel C. Sweeney, Ph.D.
Freelance Contributor to *The Audio Critic*

It appears that, despite the clinging of diehard audiophiles to pure two-channel stereo, the industry has opted for multichannel sound; therefore a hard-nosed examination of the givens is in order.

***Editor's Note:** Furtwängler was no Nazi even though he regularly conducted his orchestra under Hitler, and Dan Sweeney is no tweako cultist even though he is a former contributor to Harry Pearson's magazines* (The Absolute Sound, The Perfect Vision). *A freelance writer, especially a freelance technical writer, can't afford to regard anybody's dollar bill as tainted, not even when that bill comes from (yuck) Harry's back pocket. I have known Dan for a good many years and consider him scholarly, thorough, tuned-in, and sincere—one of the few audio journalists I respect. His knowledge of the multichannel scene is obviously wide-ranging.*

* * *

One expects that the readership of *The Audio Critic* is more than ordinarily well-informed on emerging trends in consumer high fidelity. Thus I believe I shall surprise no one by stating that the industry will very shortly be asking us to augment or replace our music systems in order to accommodate at least one and perhaps several multichannel playback formats.

Whether the reestablishment of music-oriented surround sound will be any more successful than its initial appearance in the '70s is less certain. And less certain yet is the path to be followed in adapting our carefully contrived stereo playback systems to the as yet undefined recording practices that will characterize Quad Redux or whatever one chooses to call the multichannel Restoration. This article, it is to be hoped, will provide some basis for the management of these uncertainties and for determining how, if possible, one might derive satisfactory listening experiences from multichannel music playback.

A Watershed

Heretofore this publication has had relatively little to say about multichannel playback systems. Such systems have been selling briskly, to be sure, but only as adjuncts to home-theater video installations. The music industry, for the most part, has chosen not to issue recordings in the Dolby motion-picture matrix format that is the current standard for prerecorded video software, and so multichannel could safely be ignored by a journal with an emphasis on music, and on serious music at that.

Now, with the reappearance of true, multichannel discrete formats and the announcements by a number of recording engineers that they will release music software in these formats, multichannel can no longer be dismissed—especially not in the light of the well-nigh ubiquitous perception in high-end hardware circles that multichannel formats are poised to obsolesce two-channel stereo. Thus we shall have to confront the new order, and we shall have to reexamine the basic philosophies of sound reproduction that underpin stereo and serve to define the limits of what is possible in creating the illusion of live sound.

Why Multichannel?

Except—predictably enough—for those same individuals who cling to analog records and prefer vacuum tubes, most industry professionals, whether press, manufacturers, or retailers, accept the proposition that multichannel systems are unquestionably superior to two-channel stereo in the same way that the latter surpasses mono. While relatively few industry persons have any deep understanding of human sound-localization mechanisms and how playback systems may be made to support them, most justify the assumption of multichannel superiority on the ground that adding channels somehow supplies a third dimension of depth to the reproduction.

In actuality such a rationale is just as naive as the countervailing defense of two-channel stereo on the ground that we have only two ears, and I mention it here only to steer the discussion firmly away from the fatuities which have characterized it thus far in the audio press. Neither explanation shows any real understanding of how a stereo system, regardless of its number of channels, creates an illusion of space, and neither is worth even the briefest consideration by a serious individual.

To gain such any real understanding of how a recording/playback system might create a more or less convincing simulation of an acoustical space in a real space, one must explore the field of psychoacoustics, and this I shall attempt to do—albeit cursorily—in a later installment, but before I review relevant research findings in this area, I'd like to list the perceived effects that channel multiplication is likely to provide, and then to measure these against both the claims being advanced for the new formats and the performance potential of the established two-channel format. In marketing parlance these effects are the "benefits" of multichannel, the reasons you and I will invoke should we liquidate our savings to propel ourselves firmly into the new era of perfect sound for all eternity.

The Benefits

Before we discuss the listening enhancements multichannel will supposedly provide, it might be well to examine how these formats are configured.

The new actual and proposed multichannel discrete video-sound formats (of which more presently) all provide for four or five full-frequency-range channels. The older Dolby Surround format provides for four matrix-derived channels, while the barely alive Ambisonics format can be made to yield any number of matrix-derived speaker feeds, but is usually decoded into four channels.

In video applications, the speaker arrangement for multichannel playback is invariably two or three channels across the front and one or two channels in back. For music a variety of setups has been proposed, and no real standardization is in evidence. The received opinion among manufacturers and industry pundits is that the video configuration is likely to prevail for music, however.

While generalizing about so many different formats and playback arrays can be dangerous, one can speak with a good degree of certainty about many of the basic capabilities and competences of any four- or five-channel system.

Adding a center channel between the stereo left and right will usually—depending on loudspeaker characteristics—increase the precision of imaging in the frontal soundstage. There are, however, exceptions, and I'll discuss those in a later article.

Placing an additional channel or two channels of information toward the back of the room will provide a source for ambient effects and—given the proper speaker type and placement—will allow for the precise placement of one or two sources at the rear speaker locations, though arguably nowhere else—the argument being that panning between speakers cannot be in the side and rear quadrants. Rear channels will also permit the system to maintain precise localization along the frontal soundstage while simultaneously providing for some sense of "hall"—the elusive and illusive combination of "air" and image specificity.

In addition, some evidence exists that adding channels makes the system less room-dependent. Indeed some researchers have suggested that room acoustics recede in importance as channels are multiplied, permitting the recreation of the recording space in an ordinary domestic listening space. This, however, is disputed by other authorities.

These then are the principal benefits, and there aren't so very many of them. Now to the limitations.

Four or five channels probably won't permit the extension of a sound stage to the sides and the back of the listener, at least not through simple amplitude and phase panning techniques. Although a number of individuals advocating a return to multichannel claim the opposite, published experiments don't appear to support their position.

Four or five channels won't eliminate basic problems of speaker/room interaction, such as colorations and disturbances in localization caused by early reflections and low-frequency imbalances arising from boundary effects.

Four or five channels won't provide for stable phantom images between speakers, particularly between speakers defining the back and side quadrants of the listening space. Again this assertion has been disputed by multichannel boosters, but the extant experimental evidence points the other way.

Four or five channels will not eliminate interaural crosstalk between the loudspeakers in the array.

Four or five channels will not provide an absolutely convincing illusion of depth extending behind the frontal soundstage.

Four or five channels will provide only a partial illusion of hall acoustics generally. That's because two rear or side speakers will fail to generate a completely convincing simulation of hall reverberation. In fact, according to one experiment, even six rear channels are not enough!

Four- or five-channel systems will not enable listeners to localize height.

Finally, four or five channels will not eliminate audible defects at any point along the signal chain. You'll still hear amplifiers clip and speakers resonate and whatever other forms of misbehavior may be present in the basic components.

On the other hand, four- or five-channel setups will impose much greater space demands than stereo systems, especially if identical speakers are used all round as is currently advocated by many high-end speaker manufacturers. Not only will the speakers themselves consume floor space, especially if properly situated well away from walls, but the need for unobstructed sight lines from speakers to listening positions would impose strict constraints on room decor. Of course, one could ignore such constraints, but is the perfectionist listener going to double his expenditures for speakers and electronics, and then forgo experiencing the full performance potential of his prohibitively costly system?

In the light of these very real limitations one would think that the industry, at least the perfectionist wing of it, would view the reintroduction of multichannel with some degree of skepticism, but such is not the case. In the course of a normal work week, I engage in at least three or four conversations with manufacturers' representatives pertaining to multichannel—the subject is that timely right now—and the normal industry response is not just uncritical advocacy but what I can only describe as delirious enthusiasm. Everybody is taking deep pulls from the same industry hookah, and they're all succumbing to the same unbridled euphoria. Objections such as I've raised above are blithely brushed aside with confident assertions that the experience of multichannel will be so overwhelming, even with haphazardly arranged systems, that listeners will be overawed.

Perhaps so, but my own fairly extensive experience with multichannel tends to temper my enthusiasm. During the early '80s I was friendly with a recording engineer who had an extensive library of quad mixes on four-track reel to reel—fully discrete and of very high fidelity. I heard a lot of this material, most of it consisting of recordings of small jazz ensembles where one instrument was apportioned to each speaker. The recordings were certainly vivid and undoubtedly interesting, but the illusion of a real recording space was not particularly convincing. Of course, more sophisticated recording techniques might have yielded a better approximation of reality, but that begs the question of what recording techniques are indeed appropriate. I might add that this individual was a seasoned professional with a very strong commitment to quad and a thorough knowledge of quad miking practice. Most of the individuals who will be recording in 5.1 during the next few years will almost surely be less knowledgeable.

But this is all an opinion, a matter of personal taste, and my blase reaction may be anomalous. Ultimately, you'll have to decide for yourself. And, it is to be hoped, the information that follows will assist you in doing so.

Formats

To experience multichannel playback, you obviously have to have multichannel sources. Several currently exist, though there isn't a tremendous amount of multichannel music software out there yet.

The first format might be termed incidental or accidental multichannel. Recordings made according to the MS or Blumlein techniques can be made to produce derived center and rear channels of information by means of a simple combining matrix, where stereo left and right are summed and differenced (that is, common elements are canceled out by phase inversion). Dematrixing for the purpose of deriving the four channels may be done at line level with such devices as Fosgate-Audionics surround sound decoders, or at speaker level with Panor's revival of the old Dynaco QD-1, more or less a recreation of the old Dynaquad Hafler matrix decoder.

For some persons the Dynaquad version of multichannel is all they have ever needed, but it must be noted that the approach has never won anything approaching widespread acceptance, and for at least two very good reasons.

First, there's the dearth of suitable recordings. Blumlein may be every audiophile's theoretical ideal, but most recording engineers appreciate a few more options than Blumlein techniques provide. Then there's the fact that simple Hafler decoding provides for only 3 dB of separation between adjacent channels, which means that the frontal soundstage tends to sound disturbingly monophonic as a result of the addition of an L + R center channel. Some listeners like that, but this one doesn't, nor does the listening public at large apparently.

Ambisonics

To discuss Ambisonics at all is to open an institutional-sized can of worms which I have no desire whatsoever to plumb. Nevertheless, Ambisonics still survives (to about the same extent as the California condor), and since at one time or another it has enlisted the support of some of the most eminent thinkers among audio academicians, it deserves at least a few words.

Ambisonics is the most recent incarnation of a format initially developed in Japan in the '70s and dubbed UMX in its homeland. As such, the format is in a real sense the last survivor of the quadraphonic experiment.

Never really embraced by the Japanese music lover, UMX eventually found a home in Britain where it permutated into Ambisonics (horrible name!), and such luminaries as Michael Gerzon and P. B. Fellgett contributed to its theoretical foundations and promoted it in scholarly publications. Basically an extension of MS techniques, Ambisonics used a matrixed three- or four-mike coincidental setup where an omni was juxtaposed with horizontally crossed figure eights and occasionally a single vertically oriented figure eight as well. Each mike fed a separate track on a tape recorder, and on playback the tracks were differentially combined in a matrix to produce separate speaker feeds. In the consumer B format, additional matrixing was done in the recording so as to combine the three or four tracks on the master into a stereo pair.

Ambisonics was normally played back over four similar speakers arranged in a diamond, but additional speakers could be used, and the diamond arrangement could be modified as well, thanks to various mix options on the matrix decoder.

Many technical papers have been published purporting to show that Ambisonics provides an adequate sampling of a soundfield to allow for nearly perfect recreation in a playback system. But all such analyses I have seen ignore two significant factors in the recording/playback process.

The first is arrival-time differences from ear to ear, a major sound localization cue for humans. True, Ambisonic microphone setups do encode side-to-side phase differentials, but those are not precisely the same thing, and because the setup is essentially single-point, there's no distance offset to mimic the disposition of human ears. The second phenomenon is the crosstalk components produced by pairs of loudspeaker, whereby each ear hears both speakers simultaneously. Ambisonics, like normal stereo and most multichannel formats, makes no provisions for addressing such components.

In any case, all existing consumer Ambisonic recordings have been issued only in the B format with its badly degraded separation, and thus actual playback has tended to bear the earmarks of the passive Hafler matrix—plenty of ambience and envelopment but imprecise localization.

Discrete Ambisonics could easily be accommodated on compact disc by the new AC-3 or DTS compression algorithms, but no one in the recording industry is promising anything. Currently Ambisonic recording activity appears to be quiescent, and none of the major labels ever showed any real interest in the format. Small English labels such as Nimbus were the chief software supporters, while Meridian and, very briefly, Onkyo have been the only major hardware manufacturers to offer decoders. *[See the Onkyo review on page 42.—Ed.]*

I haven't cared for the few B format recordings I've heard, but opinions differ. I would imagine that fully discrete stuff might be very convincing—much like a Blumlein recording stretched out in two dimensions.

Much Ambisonic material has been reissued on compact disc. Meridian still makes a decoder, the $3600 Model 565, which also does Dolby Pro, THX, and much else. If you feel compelled, you can still get into Ambisonics and join the worldwide fraternity of True Believers who feel it's the most significant development in the history of recorded sound. Just don't expect much new material to appear.

The Dolby Motion Picture Matrix

Here at last we're firmly in mass-market territory. As most of us are aware, the Dolby motion picture four-into-two-into-four matrix has become the standard format both for theatrical feature films and for home video releases, and indeed so popular is the format that most receivers sold today include onboard surround-sound processors for decoding Dolby matrix materials.

Less well-known is the fact that a considerable number of music recordings exist in the Dolby matrix format. BMG/RCA have released scattered titles, and Delos has put out a fair portion of its catalogue in Dolby Surround. *["Encoded naturally...through the use of microphone techniques," not with a matrix encoder, according to Delos.—Ed.]* And because the Dolby matrix uses similar encode/decode equations as the old Hafler matrix, it's entirely possible to send Blumlein recordings through a video surround-sound processor and get pretty acceptable results.

Still, I don't see that the Dolby matrix offers any compelling inducements to the music lover to expand his system to multichannel if music reproduction is the sole function of the system.

For one thing, the amount of music software, even with the inclusion of purist Blumlein recordings, isn't sufficient to provide clear justification for a multichannel makeover. But beyond that, the operation of the active matrices used in all current Dolby processors is so problematic as to invite rejection by any truly discerning listener.

At the heart of the problem is the matrix multiplier, the logic circuit used to increase adjacent channel separation in the output and thus to create a semblance of discrete multichannel sound. Such an active matrix decoder's logic works by detecting channel dominance on a momentary basis and then applying cancellation signals to the channels adjacent to that carrying the dominant signal. These cancellation signals consist of right total or left total in either normal or inverted polarity, and they serve to null crosstalk components to which they always bear an antiphase relationship. The cancellation strategy does work impressively in terms of meeting its stated goal of heightening separation, but it is not without side effects, and these side effects are of such severity as to call into question the whole notion of using the matrix multiplier for music reproduction. The cancellation signals themselves are usually generated by devices known as VCAs (voltage-controlled amplifiers), which are commonly used in signal processors of many sorts and which do just what the name implies, control an output voltage by means of an input voltage. Unfortunately, while the signal passing through the VCA is ramping up in response to the input, it is being amplitude-modulated and thus distorted. All of the distortion components thus created are passed on to the next stage, in this case to the output channels of the matrix decoder. Furthermore, since the distortion components do not bear an antiphase relationship to anything in the crosstalk, they will not be subject to cancellation.

Ever notice why distortion specs are almost never listed for surround-sound processors? Because they're terrible, at least when the device is measured on a dynamic basis, which is the only way it should be measured.

Cancellation-signal-borne distortion does not exhaust the list of decoder aberrations, however. Gain riding concurrent with cancellation is another problematic aspect of high-separation decoding. Gain riding is done to ensure constant power in a channel as crosstalk components are being canceled; naturally the cancellation process reduces the signal amplitude in the processed channel and so a boost is required. Unfortunately, like the cancellation process itself, the gain riding process creates unavoidable degradations in the output signal.

Because gain riding amplitude-modulates the signal, it produces the familiar amplitude-modulation distortions, i.e., harmonic and intermodulation distortion. But as is the case with single-band companders, which themselves are essentially automatic gain riders, the gain-riding circuits in high-separation decoders are prone to breathing effects—effects that are not always masked by the signal.

Pumping and breathing can also occur when cancellation signals suddenly stop, allowing crosstalk components to resurface. This particular embarrassment results from the fact that the logic can only address a single pair of adjacent channels at any one instant; in other words, crosstalk components are only removed selectively on a channel-by-channel basis, so that the system is not highly separated all round. Essentially the logic circuit works only on one quadrant at time, and when it is tending to one pair of channels, crosstalk is occurring elsewhere. Such crosstalk creates a distracting, level-dependent background—though in present-day decoders this problem has been much ameliorated over the norm in decoders of the past.

Of course, there are digital decoders on the market, and these dispense with the VCAs—prime sources of the distortion problem in analog implementation. Unfortunately, the digital logic does emulate the performance of the VCAs and the amplitude modulation still occurs. There are certainly advantages to digital matrix decoding—you don't have to worry about circuits drifting—but they don't eliminate the fundamental problem of the matrix, which is distortion due to gain modulation and noise due to gain riding.

Obviously, the more work the logic circuit is doing, the more breathing and distortion it is producing, and for this reason among others Dolby specifies long time constants for the logic, causing it to generate cancellation signals at fixed levels for periods on the order of a second, except in the presence of very marked shifts in channel dominance. This means that a lot of fairly low-level information is escaping the logic and is making its way through the outputs, accompanied by a great deal of crosstalk.

It must be said that when the Dolby Pro Logic processor is decoding movie sound tracks, this liability is not as bad as one might think, simply because movie recording techniques have grown up around the matrix and its limitations, and more than likely Dolby personnel have supervised the mix to avoid taxing the capabilities of the circuit. Most movie-sound recordists just don't place important information in all channels simultaneously, so the crosstalk, when it occurs, isn't ordinarily going to compromise imaging very much. But with complex musical scores one can argue that the Dolby Pro Logic time constants aren't ideal. From my experience, the Fosgate decoder circuits based on variable time constants do a better job on music.

But all this is very much beside the point, really, because the music industry shows no signs whatsoever of adopting the Dolby matrix. No major artist or recording engineer supports it, and all attempts to build a niche market for surround-music playback among high-end buyers have failed. The matrix will undoubtedly remain the norm for movies for some time to come, but its prospects in the music business remain dim.

THX

THX is not a multichannel format as such. I mention it here simply because it is often misidentified as a format and because the THX program does play a very significant role in the multichannel universe.

THX began as a motion-picture theater certification program and as a means of promoting the ideas of Tomlinson Holman, founder of the program at Lucasfilms, for improving theater sound systems. The consumer version of THX, which was developed much later, is essentially a licensing program for components and, latterly, software to be used in multichannel home-theater systems.

THX standards cover laserdisc players, laserdiscs, surround-sound processors, amplifiers, speakers, screens, and even interconnect cabling. The program endorses both the Dolby matrix and Dolby AC-3 discrete 5.1-channel formats, and all THX-licensed processors must meet Dolby specifications.

It is not my intention to discuss the THX standards here except in regard to the processor. (A fuller consideration of the program will be reserved for a possible later article.) At least one aspect of the processor standards is very germane to the present discussion, however, that being the TXH decorrelation circuit, which has a direct bearing on the role envisioned for the back channel by Tom Holman.

Normally the surround channel in the Dolby matrix is reproduced over two loudspeakers placed in the rear or along the sides of the listening space, a single speaker having been found to produce undesirable localization effects in the presence of diffuse, ambient information. Holman feels that the use of two speakers, absent appropriate processing strategies, merely ameliorates the problem—in other words, one still tends to to localize sounds to the speaker location, only now one perceives two speaker locations rather than one.

Since Holman has always maintained that the reproduction of ambience constitutes the paramount purpose of the rear channel, he finds localization to the speaker position to be a significant problem. He has proposed solving that problem by altering the signals so as to muddle directional cues. The best solution, Holman feels, is to make the signals reproduced by the two surround speakers slightly different from each other, and to that end he developed a technique he calls, somewhat confusingly, pitch shifting, whereby different equalization curves are imposed upon either channel. This type of decorrelation was present in all THX-approved processors initially, though later phase-shifting circuits were approved as well.

In a full THX system, the decorrelated signals are always reproduced by dipolar rear speakers, where the listener sits in the null of either dipole, that is, sits directly beside the speaker so as to perceive almost no direct sound. The reflected, decorrelated sound he does perceive is supposed to reproduce ambient effects more successfully than the output of conventional systems, such output consisting of unprocessed direct sound.

Assessing the ultimate success of the THX treatment of the rear channels is not easy. From my experience it is indeed difficult to localize to the rear speakers in a THX system—and I've reviewed several for other publications—but I've always felt that there was something not quite kosher about the effect. It's not an effect you'd likely hear in any other setting, and so it seems vaguely unnatural—at least to me it does. Lucasfilms have performed many listening tests purporting to prove the efficacy of the THX methods, but there isn't a lot of independent research to back up the THX positions. I'd say the jury is still out.

I'd note here that THX notions concerning decorrelation and diffuse dipolar rear speakers were conceived to support the Dolby matrix with its monophonic surround channel, which conveys difference information. Many questions have arisen regarding the suitability of the THX speaker configuration for the 5.1 discrete formats. (Decorrelation is obviously unsuitable for two discrete rear channels.) The answers to those questions will come in time when 5.1 software becomes more widely available, but at this time the evidence to form firm judgments is simply insufficient.

The DTS 5.1 Format

DTS (Digital Theater Sound) is a Southern-California-based company owned in part by MCA-Universal. The company's chief business is the licensing of their proprietary data-compression technology and the sale of audio playback equipment to the motion-picture exhibitor industry, but the firm also has eyes on the consumer audio and video markets, where it hopes to establish its six-channel system as the new standard.

DTS is hardly a household word, but the company itself has to be considered a comer. The DTS digital sound system for movie theaters, which was first used in Steven Spielberg's *Jurassic Park,* is today employed in many more theaters than the rival Sony SDDS or Dolby SR-Digital systems—over 5000 screens at the time of this writing, and the company appears poised for eventual dominance in this market.

Nevertheless, in spite of DTS's remarkable success in pro audio, the good industry position within the consumer electronics press and among most of the audio hardware manufacturers is that DTS is a negligible and laughable presence in the consumer electronics industry and that, conversely, Dolby's rival AC-3 consumer system has already emerged as the clear victor in the multichannel format war.

Conventional wisdom on competing technologies is nearly always correct because it reflects industry consensus and thus forms the basis of a self-fulfilling prophecy, but it seems to me that in this instance it cannot be entirely trusted. A careful consideration of the current positions of DTS and Dolby simply does not warrant the assumption that DTS can be dismissed out of hand. MCA's resources exceed those of Dolby by a wide margin, and DTS's growing dominance within the film industry has to be counted as a factor, particularly when one considers that Dolby's 5.1-channel digital theater sound system is dead last behind Sony SDDS. Bear in mind also that Dolby's celebrated triumphs, i.e., winning the endorsement of the Laserdisc Association and the HDTV Grand Alliance may not be decisive, since the issue of what system will be adopted for the mass-market digital video disc is still outstanding. Toshiba-Warner is offering lukewarm support for the Dolby system while providing room on the discs for simultaneous DTS audio tracks, while the Sony-Philips consortium has yet to announce a standard. Until DVD standards are completely resolved, DTS is still in contention. *[This was written before the Toshiba-Warner and Sony-Philips factions finally agreed to support a single DVD standard.—Ed. J*

Concerning the format itself, the DTS consumer system has been configured to produce a "5.1 channel" output, meaning that five fully discrete full-range channels are specified along with one subwoofer channel. This configuration is already standard for digital movie theater systems, and since the initial market for discrete multichannel is expected to be in the area of video software, movie-theater practice will prevail in the home as well.

Because DTS must fit six channels of information into the space occupied by two channels of compact-disc-video audio, the signals for each of these six channels must necessarily be compressed; that is, data must be selectively removed—hopefully in such a manner that deletions will always go unnoticed because of the masking effects of the program material. However, compression in the DTS system intended for laserdisc and the compact disc is relatively gentle as compression goes, occurring at a 4:1 ratio, which means that one quarter of the initial quantity of data is preserved. While that may seem pretty severe, be aware that the Dolby AC-3 consumer system operates at an 11:1 compression ratio, while the Dolby SR-Digital theater system is even more severely compressed.

The issue of compression is the focus of much controversy in the audio press, with subjectivists—not unexpectedly—being against it, and the *Stereo Review* crowd being very much for it. Since nobody in the press has much, if any, experience in comparing compressed with uncompressed material, the positions are essentially ideological.

Still, no one can argue convincingly that compression can remain transparent at any degree of severity. At some point data losses will become audible, but at what ratio? Twenty to one? Ten to one? Four to one? Two to one?

Moreover, the issue is vastly complicated by the existence of several competing compression algorithms, all of which are deemed superior by their respective backers. "We can get away with 10:1, while they sound lousy at 4:1" is representative of manufacturers' claims in this regard.

So is DTS at 4:1 superior to Dolby AC-3 at 11:1? Not necessarily. If Dolby's algorithm enjoys the edge its backers claim for it, then the Dolby system could outperform the DTS system with less than half the data. But we don't know if Dolby's algorithm really is better because we lack the evidence to make such determinations.

Now it would seem that a listening demonstration, where signals from master recordings were submitted to either scheme and the results compared, would go a long way toward resolving the issue, and Terry Beard, the president of DTS, has told me personally that he would submit his system for such comparative evaluation at any time. However, Dolby has allegedly declined all such invitations and will not submit its system for comparison even to the companies backing digital video discs. Dolby representatives have indicated that the company has performed internal listening evaluations, and Dolby management is satisfied with the results.

In any event, the Dolby format is already available on laserdisc, and numerous titles have been issued as of this writing. But thus far not one piece of music programming has been issued on compact disc.

On the other hand, no video material has been released in the DTS consumer format, but two compact discs have been issued, on the *dmp* label and on Brad Miller's Mobile Fidelity International. Both Miller and *dmp's* Tom Jung have indicated that they will releasing more material in 5.1 in the future.

Dolby AC-3 5.1

Like DTS, the Dolby AC-3 format is a compression algorithm, not a channel allocation per se. Unlike the DTS algorithm, Dolby's has been described in detail in white papers available for the asking from Dolby Laboratories and in technical articles in the consumer press.

If you haven't already steeped yourself in the background material surrounding the Dolby vs. DTS controversy, I'm sure you're asking yourself why Dolby would opt for a higher compression ratio since, at best, maintaining transparency then becomes more difficult. The answer is that Dolby did so at the behest of two industry committees, the one controlling the destiny of high-definition television in the U.S. and the other established to set standards for the 12-inch video laserdisc.

In both cases the capacity for wideband multichannel audio was limited. In the case of HDTV, the limits were set by the video requirements and the spectrum allocation—and there wasn't much way around either. But in the case of the laserdisc, the problem was a bit more complex.

The Japanese wise men who set the course for the consumer mass media decided that backward compatibility with the existing laserdisc must be maintained as much as possible. That meant preserving one of the analog tracks and both of the stereo digital tracks. Which in turn meant that all of the data for the six new audio channels had to go in the space taken from a single analog track. And that, my friends, is a tight squeeze.

DTS proposed a different solution, one unacceptable to the Laserdisc Association. Why not scrap the two stereo digital channels and leave the two analog channels? All laserdisc players can play FM analog, so the old players can play the new discs, albeit in analog mode only.

Association leaders were quick to object that the analog sound was inferior and that customers would not be happy. They also could have argued that in some parts of the world, such as the U.K., the newer players cannot in fact play analog at all, and thus if DTS were introduced as a worldwide standard, it would require that foreign retailers maintain a double inventory of discs.

The issue is further complicated by the fact that the Japanese are nurturing hopes that the new 5.25-inch video disc will do what the laserdisc never could and win a mass market. Since compatibility is not a problem with the little disc, severe compression is scarcely required. And of course if the little disc becomes a big seller, the continued allegiance of a few Japanese firms to the laserdisc format will surely be reexamined. And if laserdisc is on the way out anyway, why push so hard for heavy compression? Reserve it for HDTV broadcasting and leave it at that.

Of course, it might be objected that severe compression is highly desirable, if for no other reason than it gives an opportunity to the rationalists in the industry to stick it to audio tweakdom bigtime. You didn't like the compact disc with 20 kHz bandwidth and 16 bits? Well, how about an average bit rate of 8 bits? Put that in your pipe and smoke it. Whether such a motivation is going to figure in industry deliberations is somewhat doubtful, however.

Describing Dolby's adaptive subband coding technique in any detail would constitute a lengthy article by itself, so I'll confine myself to the bare essentials here. Two basic principles are embodied in the coding—selective reduction of resolution on a sample-by-sample basis reflecting spectral distribution of signal content, using bandpass filters to control granulation noise attendant upon bit reduction, and assignment of bit allocations based on masking principles. Like the analog spectral recording circuit to which it is somewhat related in principle, the Dolby AC-3 encoder looks at the signal from moment to moment, decides where noise will be masked by the signal content in contiguous and adjacent bands, and allows the noise to occur where the signal will bury it. In addition to subband coding, the Dolby AC-3 uses delta modulation to effect a further considerable reduction in bit rate. High-frequency channel separation is also selectively reduced where it is deemed to be inaudible.

Unlike DTS's Terry Beard, Dolby representatives have never claimed that their system is completely transparent—at least not at the specified 11:1 compression ratio. They simply maintain that it represents an improvement over the Dolby matrix—a position that would prompt little disagreement in the industry.

Dolby AC-3 can be decoded into four, two, or one channel as well as in the 5.1 format, but it will probably appear in music software in 5.1. Thus, as with DTS, music recording will mostly likely hew to a pattern best suited to public address applications within commercial movie theaters.

[This introductory article does not cover the specifics of recording for multichannel and of optimizing the listening environment—speakers, rooms, etc.—for multichannel playback. Further articles by Dan Sweeney on these and related subjects are in the pipeline and will be seen in future issues.—Ed.]

Loudspeaker Systems

(continued from page 19)

less forbiddingly priced design. Snell Acoustics would be doing all of us a favor by making such a speaker happen.

[Flash! The above was on its way to the printer when I learned that Kevin had left Snell Acoustics and joined Harman International, where his mandate will be to create an "ultimate" speaker system for a newly formed division of the company. Snell has new owners. End of an era.]

Snell Acoustics Type MC LCR 2800 and SUR 2800

Snell Acoustics, Inc., 143 Essex Street, Haverhill, MA 01832. Voice: (508) 373-6114. Fax: (508) 373-6172. Music & Cinema Reference System: MC LCR 2800 center-channel loudspeaker system, $4799.00 each; MC SUR 2800 dipole surround tower, $6198.00 the pair. Review samples on loan from manufacturer.

These models are intended to complement the Type MC Reference Towers in the THX-certified Snell Music & Cinema Reference System, but they will work just as well with the non-THX Type A towers. I received them a very long time after the A's, so their testing was inevitably delayed. You can expect detailed reviews in the next issue. I have one important observation to make here, however.

The SUR 2800 would be a hazard in households with small children, dogs, or physically uncoordinated old persons. It will almost surely come crashing down if bumped hard because it is well over 7 feet tall, has a footprint about the size of the magazine you're reading, weighs 110 pounds, and its center of gravity is *way* up there. Yes, a pair of them will launch the surround sound over everybody's head without any obstacles, but to my mind they represent the triumph of engineering rigidity over common sense.

I'm almost certain I'll have nicer things to say about their performance in the review.

Coming:

- Reviews of some very interesting loudspeaker systems, including the new Waveform Mach 17, the NHT Model 2.5, the AR Model 303, and others.
- The next installment of our FM tuner survey.
- A first look at Bob Carver's Sunfire "True Subwoofer," an 11-inch cube claimed to reproduce 18 Hz at 110 dB SPL with low distortion.
- More reviews of multichannel AV electronics, including some with AC-3.
- A reevaluation of the MiniDisc in the light of third-generation Sony technology.

AV/Ambience/Surround Equipment: Our First Look at Multichannel Processors

By Peter Aczel
Editor and Publisher

Is two-channel stereo obsolete? Just in case it isn't, these multichannel front ends have perfectly adequate stereo modes, but the question is—are they already obsolete for multichannel?

Testing electronic signal paths in audio equipment used to be a fairly obvious, even if sometimes complex, procedure because all the tests were basically about one thing—does the output resemble the input? The new generation of multichannel AV electronics changes that perspective totally—the output is a highly processed and therefore completely altered version of the input. Only the front left and front right channels follow the basic rules of stereo and even they are almost invariably routed through the processing stages without a bypass, processed or not. That makes testing from the point of view of the audio purist not only highly problematic but, in the final analysis, irrelevant.

One could, of course, try to verify whether or not the processing follows accurately the prescribed protocols of Dolby, THX, or what have you, but even that isn't necessarily a measure of quality when the processing is implemented with standard ICs that everybody uses. Furthermore, new formats with new protocols are coming—AC-3 has quietly arrived already (without much AC-3-encoded software, to be sure)—so that the models reviewed here may have a short life expectancy at this point, although they and others like them are what's currently out there. Their input/output accuracy appears to be limited in nearly every case by all the extra silicon in the signal path; forget about anything even approaching 0.001% (-100 dB) distortion; -80 dB is more like it.

For all of the above reasons, the reviews that follow are not our usual fine-tuned critiques of electronic performance but merely brief evaluations of engineering thinking, circuit design, parts quality, and user interface. I recommend that you first read David Rich's description of the Mcintosh C39's design details in his sidebar to the feature article on Mcintosh Laboratory, further back in this issue. Then come back here and continue. If you are totally unfamiliar with, or even just slightly confused about, the various multichannel formats implemented by these procesors, then the Dan Sweeney article on the preceding pages is required reading before you get involved in the reviews.

B&K AVP2000

B&K Components, Ltd., 2100 Old Union Road, Buffalo, NY 14227-2725. Voice: (716) 656-0026 and (800) 543-5252. Fax: (716) 656-1291. AVP2000 Multi-Zone Remote Controlled AW Preamplifier with Dolby Pro Logic, $998.00. Tested sample on loan from manufacturer.

This is both an amazing and a frustrating piece of equipment. Amazing because its circuitry is engineered with the utmost sophistication and elegance to make it an up-to-the-minute, ultrareliable, full-featured AV control unit; frustrating because the user interface is so unfriendly as to evoke a "forget about this damn thing" reaction from the average user. Luckily the control settings in a typical home-theater installation are left pretty much alone, or at least changed infrequently and even then not radically. Memory presets also help to some degree.

No on-screen displays are available. All control functions depend on the 5-inch fluorescent display on the front panel, which can display 16 alphanumeric characters in a single line. You step through menus with subdirectories, one line at a time, select the function you want, and press Enter. You do this by means of the remote control or, with fewer options, a row of buttons on the front panel. The displays are cryptic and too small to be read from a normal viewing/listening distance, unless your vision is far better than 20/20. If you misplace the remote, you lose some of the functions (such as, for example, balance). I became irritable using the B&K for more than a few minutes; techie geeks probably won't mind.

As there is a streak of techie geek in me, and more than just a streak in David Rich, we loved, on the other hand, the simplicity and rationality of the circuit design. For example, there are no pots anywhere in the signal path. "This thing will last forever," David wrote in a note to me. Both the volume and the balance functions are implemented with the Sanyo LC7535 passive programmable attenuator. Only one stage is required for these function because the LC7535 is a logarithmic device that changes the level in 1 dB steps. All audio switching is in CMOS logic, using the Sanyo LC7821 chip. Zone 1 and Zone 2 (what Mcintosh calls Area A and Area B) each use only one op-amp, an Analog Devices OP249, in the audio signal path, and there are no coupling capacitors. Now that's a truly clean approach, the kind that gladdens the heart of the audio purist; the trouble is that THD + N measurements still run into what would appear to be the AV-processor "stone wall" of -80 dB (0.01%); in this case the curves for the front outputs bottom out at -76 dB to -78 dB, almost regardless of frequency The Sanyo chips just mentioned, which are always in the signal path, are the most likely limiting factor here. At all but the highest output levels there's about a 13 dB drop in noise, though, when all the Dolby processing hardware is bypassed ("direct" mode). With the inputs shorted (zero signal), the noise floor in the direct mode is 20 to 25 dB lower than when the Dolby circuitry is in the signal path. Maybe it can't be done any better because all these figures are well within the published specs. One respect in which the B&K AVP2000 is fully the equal of top-notch stereo-only preamps is crosstalk; in the direct mode, at full gain, front left/right out, I measured -100 dB at the lower frequencies, -95 dB at 1 kHz, and a maximum of-70 dB at 20 kHz (ref. 2.2 V out).

How does the AVP2000 compare to the Mcintosh C39, our conservative/comprehensive high-end archetype in this product category? The build quality is about the same, at 28.5 cents on the dollar. Of course, there's less stuff in the B&K—fewer inputs, fewer features (e.g., no phono), no separate record bus (you have to record what you're listening to), less protection, no user-friendly knobs, as I said; on the other hand, B&K scores with an unexpected quality touch here and there, such as for example ±15 V on the analog power rails as against ±12 V in the Mcintosh. The surround processor board is the same in both units, except that the B&K uses cheaper TL071 op-amps for the subwoofer summing and all filtering; however, filter orders and frequencies appear to be identical. THX costs $700.00 extra (B&K AVP4000).

So, the decision regarding the purchase of a B&K AVP2000 hinges mainly on the acceptability of its user interface. The unit is AC-3 upgradable, so that issue shouldn't stop you. The basic engineering is superb.

Lexicon CP-3 PLUS

Lexicon, Inc., 100 Beaver Street, Waltham, MA 02154-8425. Voice: (617) 736-0300. Fax: (617) 891-0340. CP-3 PLUS. Digital Surround Processor with remote control, $3200.00. Tested sample on loan from manufacturer.

Depending on your priorities, this is either the most desirable surround-sound processor extant or an expensive luxury you can do without. As an all-purpose AV control center for an elaborate home-theater system, the Lexicon *CP-3PLUS* is definitely underendowed. As an engine for sophisticated multichannel audio processing in a domestic chain, it has no equal. For example, it has only four inputs for program sources, only one tape output, and no S-video facilities; on the other hand, it can drive seven audio channels plus a subwoofer channel with an almost endless variety of surround/ambience/reverb effects, all of them digitally produced and far beyond the capabilities of other processors.

Unfortunately, Lexicon is a somewhat nervous outfit when it comes to lending equipment to reviewers; we received no circuit schematics to be analyzed by Dr. Rich and were pressured to return the *CP-3PLUS* before we had done as much with it as we would have liked to. Thus my comments here are necessarily brief. I call your attention to David Ranada's exhaustive user evaluation in the May 1995 issue of *Stereo Review,* I was not able to get as far with this equipment as he did—and I trust his critical skills.

The user interface of the Lexicon bears a definite resemblance to that of the B&K discussed above (despairingly), but it is friendlier in several respects. There are two lines, not just one, of 16-character alphanumeric LCDs on the front panel, and more importantly, the processor has a character generator for a video overlay display on a TV screen. Yes, everything is menu-driven, and the menus are labyrinthine, but the on-screen display shows the full menu in each case, not just a fragment of it, and that makes all the difference in the world. I still like knobs and switches (one of my old lab instruments from the 1970s has 54 of them), but I realize that Lexicon's implementation is the logical one when there is such an overwhelming wealth of functions and settings.

I won't even attempt to go into the details of what the *CP-3PLUS* can do—the manual is humongous—but it's all done with DSP, which is by far the most accurate way, not with analog chips (and that goes for Dolby Pro Logic and THX, as well as the various Lexicon proprietary functions). When I tried the *CP-3 PLUS* as the front end for my reference stereo system, however, I wasn't particularly pleased; I often need more than four inputs and have little patience with long menus when I'm used to turning a knob by two clicks for a particular function. The sound was perfectly fine, of course. As a control center for my home theater system, I found the Marantz

AV600 to be more convenient, more video-friendly, easier to set up, and easier to use under standard conditions (especially when shared with the whole family), though not nearly as sophisticated in terms of technical features.

As for measurements, I found some peculiarities. For one thing, I was unable to set the unit for dead-flat frequency response. With input and output at unity gain, the best I could get at the front left/right outputs was a gently falling response below 1 kHz, namely -0.05 dB at 400 Hz, -0.1 dB at 100 Hz, -0.15 dB at 20 Hz, -0.34 dB at 10 Hz. Not very significant, but a Bryston it ain't. Above 1 kHz the response was flat up to 20 kHz and dropped to -0.15 dB at 40 kHz. THD + N once again ran into the -80 dB "stone wall" of the surround-sound world; that's exactly what I measured at all frequencies at the maximum front output of 7 V, just before clipping, with input and output at unity gain. Changing the load to 600 ohms affected only the 20 Hz minimum disortion (-75 dB) and reduced maximum output to between 4 V and 5 V. With input and output at maximum gain, THD + N increased by 10 dB to 15 dB! The noise floor of the Lexicon with shorted inputs was comparable to that of the B&K AVP2000, certainly not lower, perhaps even higher here and there by a couple of dB. Front left/right crosstalk was highest, -74 dB, at 15 kHz and dropped steadily with decreasing frequency to -112 dB at 20 Hz; very good indeed. That was as far as I went before I had to return the unit.

My general assessment of the *CPSPLUS,* without the benefit of a circuit analysis by David Rich and further study, is that it is not for the typical audiophile user, even if he/she can afford it. It is basically a piece of professional gear, albeit entirely compatible with domestic systems. That it has some awesome capabilities is without question.

Marantz AV600

Marantz America, Inc., 440 Medinah Road, Roselle, IL 60172-2330. Voice: (708) 307-3100. Fax: (708) 307-2687. Model AV600 Preamplifier/Tuner, $1199.99. Tested sample on loan from manufacturer.

"All theory, dear friend, is gray," Mephistopheles tells Faust, and the Marantz AV600 proves it. Of all the AV control units reviewed here, the Marantz is "theoretically" the least desirable, but it is the one that I left in my home theater system because of its practical advantages. A clean, elegant design is definitely not what it is; with the surround-sound decoder and the subwoofer filter defeated, the signal still passes through 11 active stages plus 16 blocking capacitors, 7 wiring harnesses, 4 CMOS switches, 3 analog pots, 2 digitally controlled IC pots, and 1 relay, in order to get from the input to the output (David Rich's count). On the other hand, it has every conceivable function, feature, and convenience you could ask for in the front end of a home theater system, and they all work. On top of it, the user interface is friendlier than most.

Even an FM/AM tuner is included in this design in its pursuit of all-inclusiveness but it's a rudimentary one; we didn't even bother to test it for our survey of high-quality tuners in this issue because the specs put it in a totally different category (image rejection 50 dB, other specs also of entry-level standard). The IF, FM demodulator, stereo demodulator, and AM radio circuits are all in one combo chip, the Sanyo LA 1856 (strangely, they do not use the Philips chip set). Overall, it's a mid-fi type of tuner design.

The build quality of the AV600 is similar to that of other Japanese equipment in its price range—single-sided PC boards with jumpers, etc.—and not as good as that of the U.S.-made B&K and Mcintosh units. The preamp section has no phono, and those 11 active stages use el cheapo JRC (New Japan Radio) NJM4558 and NJM2058 op-amps. Input-level and tone controls are likewise cheap analog affairs instead of solid-state controls that offer greater reliability. The tone controls are not defeatable. One nice feature is that a stereo pair of subwoofers can be accommodated, with 80 Hz fourth-order lowpass filtering. The switchable 80 Hz highpass filter design for the front and center channels is second-order. Sallen-and-Key circuits are used for all filters. The Dolby Pro Logic decoder is also from JRC, an NJM2177. THX is included in the base price; the decoder uses the Analog Devices AD 1877 for A/D conversion and the Sanyo LC78835 for D/A; the DSP engine is the Yamaha YSS205B, which also does the delay functions for Pro Logic. In the mono and stereo modes, the surround decoder is bypassed. The power supply is a pleasant surprise, as it provides a full ±15 V (even the Mcintosh C39 is limited to ±12 V).

The superiority of the user interface to what we usually see in AV control centers is due to (1) very good on-screen readouts and (2) an ergonomically superior and very comprehensive remote control. Unfortunately, the on-screen display capability is disabled when the S-video connections are used, so that if you want the better color performance available with S-video, you are reduced to the 10-character alphanumeric display on the front panel (which is well-designed and surprisingly informative). This is an AV tradeoff which should disappear as more sophisticated equipment comes on the market.

The somewhat heavy-handed circuit design doesn't appear to affect basic performance, at least not much. The "stone wall" of -80 dB (0.01%) THD + N I keep coming up against in AV processors is once again evident; 20 Hz and 1 kHz just before clipping, at 5.5 V out of front left, are at exactly that irreducible minimum value, with 20 kHz bottoming out at -75 dB and 2 V. Par for the course, almost regardless of topology. The frequency response suffers a bit on account of the unbypassed tone controls; in the less good channel I measured a bump of +0.35 dB at 80 Hz and one of +0.47 dB at 15 kHz, both impossible to get rid of with any kind of knob twiddling. But what's a fraction of a dB in that highly processed sonic soup? Front left/right crosstalk is in the -77 to -69 dB range up to 1 kHz, gradually worsening (by as much as 20 dB) on the way up to 20 kHz. That's still OK but not great. The irreducible noise floor of the unit, with shorted input, fluctuates spectrally between -110 and -95 dB as referred to 2 V out, which is again OK but could be better.

Bottom line: at $1200, with THX and all other goodies included (except AC-3), this is good value and unquestionably does the job. Furthermore, the AV600 is designed to mate perfectly with the Marantz MA500 monoblocks, still our recommendation for top value. Maybe Marantz's Japanese engineers are right—put in every possible feature, make sure everything works, make it easy for the user, and let the David Riches of the E.E. world grumble about the circuitry. Even so, check out the Onkyo below before you make up your mind.

Mcintosh C39

Mcintosh Laboratory, Inc., 2 Chambers Street, Binghamton, NY 13903-2699. Voice: (607) 723-3512. Fax: (607) 724-0549. C39 Audio/Video Control Center, $3500.00. THX-M optional THX processing card for C39, $500.00. Tested sample on loan from manufacturer.

The review of this highly elaborate but not quite up-to-the-minute unit appears elsewhere in this issue as part of the feature article on Mcintosh Laboratory.

Multichannel "Receiver"

Onkyo TX-SV909PRO

Onkyo USA Corporation, 200 Williams Drive, Ramsey, NJ 07446. Voice: (201) 825-7950. Fax: (201) 825-8150. Integra TX-SV909PRO Audio/Video Control Tuner/Amplifier with remote control, $1880.00. Tested sample on loan from manufacturer.

This remarkable piece of equipment throws in seven (7) perfectly good power-amplifier channels with the control/processing functions offered by the units reviewed above and hardly charges you extra (well, very little extra). What's more, Onkyo does it all with DSP, in the manner of the pricey, ultrahigh-tech Lexicon. Hard to resist, wouldn't you say? It looks like a trendsetter.

Now, it should be made very clear that this is no longer the current model. For a while it coexisted with the newer TX-SV919THX ($2000.00) and was then dropped from the line, probably because there was little or no interest in its Ambisonic decoding capability (see the Dan Sweeney article in this issue), whereas the THX capability of the newer model became paramount. The two models are so similar in concept and architecture, however, that you can consider this review to apply to both in all essentials. There may even be some samples of the older model languishing in Onkyo dealers' stockrooms. Since an AC-3 version is a reasonable expectation, both models may soon be superseded; the home-theater industry is still in its infancy and quite volatile.

The TX-SV909PRO is probably the most complex piece of electronic gear for home use we have encountered so far. One could spend months "deconstructing" and testing it; Onkyo must have worked on it forever. The basic picture is this:

The control section is a digital labyrinth, with DSP chips, microprocessors, A/D and D/A converters, CMOS logic, etc., ad infinitum. David Rich can parse it and calls it "a remarkable achievement;" my eyes glaze over. The build quality is upper-middle-class Japanese, better than Marantz, all neatly mating plug-in boards, no wiring harness. The power-amplifier channels, on the other hand, are quite rudimentary albeit well-designed, with only a single output device per rail. The tuner is again of the upper-middle Japanese receiver grade, better than the one in the Marantz AV600, but not interesting enough to be included in our FM tuner survey. The phono section is a single-stage circuit using a low-cost op-amp. The power supplies for the low-level analog circuits operate at ±12 V. One baffling aspect of the design is that the available line-level outputs are potted-down versions of the power-amplifier outputs (why not their inputs?). That mixes the power-amp distortion and noise into the line output. Go figure. From input to output, the signal path in the Onkyo comprises 5 low-level active signal blocks, plus the A/D, D/A, and power amp, plus 7 electrolytic coupling capacitors end to end, plus a 7-gang analog pot. That's still a lot less congested than the Marantz (see above).

The user interface is far from simple (how can it be?) but is greatly enhanced by comprehensive on-screen displays—not disabled when S-video is used, in contrast to the Marantz—and by an economically well-designed remote control, most of whose functions are duplicated by a large assortment of buttons on the main unit.

As for the -80 dB "stone wall" of distortion, this is the first piece of AV surround equipment I have tested that breaks it; THD + N bottoms out in the -82 to -85 dB range, regardless of frequency, with the power amplifier always in the loop! Clipping occurs around 140 watts into 8 ohms and 220 watts into 4 ohms. The D/A conversion (NEC µPD6376 dual-channel DACs, fed with NPC SM5840EP digital filters) appears to be limited in resolution by the analog distortion. Gain linearity errors exist from -60 dB down; the worst is -2.7 dB at -83 dB.

None of the above changes our basic conclusion that what we have here is an early blueprint of the future.

FM Tuners: The Present State of the Art of FM Reception

By David A. Rich, Ph.D.
Contributing Technical Editor

FM appears to be the orphan subject of audio reviewers, probably because most audio reviewers have no grounding in radio-frequency (RF) technology. This article is intended as a remedy.

Editor's Note: *Once again, Dr. Rich has come up with a tutorial type of survey article that may be a bit too technical in spots for some of our readers. Once again I say, don't worry about it. The consumer knowledge he imparts is clear and simple at all times. If you follow the circuit discussions, fine; if you don't, you'll still know what are the corner-cutting solutions, what a quality design entails, what to look for when you pay a lot of money, how to be an enlightened purchaser of FM tuners. It would be wrong to start running for cover as soon as you see an engineering-oriented paragraph. No other audio publication gives you comparable insights, so take advantage of them. Technical concepts have a way of sinking in, even when you find them bewildering at first. All it takes is the desire to know. Of course, as I've pointed out before,* **The Audio Critic** *is not "My First Book of Electricity." If you know the difference between a volt and an ohm, between ac and dc, you'll get something out of David Rich's techie marathons.*

* * *

The importance of FM radio.

Why are we running a major article on FM tuners? FM tuners deliver a signal that is not as good as a CD, so who needs them? Music lovers need them because they offer a world of music not available from a personal CD and record collection, no mater how large. Each week, commercial and noncommercial FM stations present "live on tape" performances of this country's symphony orchestras. They also offer other live-on-tape performance series of opera, chamber music, and acoustic jazz not available from any other source but FM radio. Commercial recordings are made only of the best-known performing organizations, and they do not capture the remarkable quality and range of the musical groups throughout the nation.

Each weekday, NPR's *Performance Today* fills part of the gap left by commercial recording companies. Live performances recorded throughout this country are presented, along with insightful commentary. The quality of the programs' production and the music itself are unmatched by anything to come from the BBC or European radio. FM radio also brings educational programming, produced by NPR, PRI, and independent sources, that helps the young and not so young understand fine music. These programs include *Adventures in Good Music, Pipedreams, The Record Shelf, Saint Paul Sunday, Sunday Opera* (with Wayne Conner), and *Schickele Mix.*

Other examples of quality music and spoken-word programs that come through a tuner include *Jazz from Lincoln Center, New Sounds, A Prairie Home Companion, Selected Shorts, Car Talk, Wade in the Water, Music from the Hearts of Space,* and *Thistle and Shamrock.* Yes, a lot of trash is on FM, but the preset switches on your tuner take care of that problem

Unfortunately, this is not the best of times for radio. Commercial pressures are pushing fine classical and jazz stations off the air. Noncommercial radio stations have started to play rock music (they call it "world music" and "adult alternative") in an attempt to boost listenership and thus increase contributions. The problem here is not that noncommercial radio systems are starved for cash but that the present distribution method, where each town has its own NPR station, is so inefficient that little cash gets used to produce programs and broadcast them. The money instead flows into the pockets of those who run the local station.

All these upheavals on the FM dial may cause you to go in search of a new tuner. For example, in Philadelphia the NPR station switched to all-news to gain listeners. The music programs are now only available on a station 40 miles away, and to receive that station well Philadelphians suddenly need a good antenna and tuner.

FM has some fundamental limitations that could be overcome by newer technologies. For example, digital radio could be transmitted across the country using a satellite. It would be similar to the DSS television system. Hundreds of different channels would be available. This is a much more efficient method than the present system. Narrowcasting is possible because the signal goes to the whole country. The satellite dish could be as small as two inches and it could even fit on a car. The technology exists today, but commercial radio stations are blocking its implementation. They fear that this technology would supersede them. Since they have been given their licenses to the FM band for free, it makes no sense that they have any right to block a better technology. Unfortunately sense is not something reliably found in the halls of government.

The fundamentals.

Enough of this philosophical discourse on radio; let's get on with the business at hand. How does an FM receiver work, and how do you know if you have a good one?

The first thing to understand is that the tuner has to do two distinct jobs. One is to extract the desired signal from all the signals and background noise it is receiving. The other is to demodulate the signal so the two channels of audio information can be recovered. The circuitry should reject as much of the noise and undesired signals as possible, so that the tuner can perform its second function as well as possible.

Different signal conditions put different requirements on a tuner. If we are trying to receive a strong signal from a directional outdoor antenna which is free from multipath (reflected signals) as well as from strong adjacent (200 kHz away) and strong alternate (400 kHz away) signals, then the tuner's first job is easy. How good the station sounds will then depend on how well the tuner demodulates the signal (Job Two). Traditional audio measurements (signal-to-noise ratio, distortion, and frequency response) tell us how well the tuner is doing its second job. The only distinction with tuners is that the measurements should be across a range of antenna signal levels, since the tuner's performance degrades as the signal level decreases. Often specs are given only at unrealistically high antenna signal levels. Accuphase, a Japanese high-end manufacturer, is unique in presenting the complete information, and they even guarantee the specifications! Of course, the Accuphase tuner costs $2995.00, so it had better perform well, and the company should have no reason to hide anything by giving only minimal specifications.

Now consider the case where we are trying to receive a weak signal accompanied with strong adjacent and alternate signals. This signal is being received on an indoor antenna having a small gain and poor front-to-back ratio (see antenna article this issue), so the signal is noisy and corrupted with multipath. Now the ability of the tuner to do Job One is most important. As we shall see below, removing the adjacent- and alternate-channel signals is going to distort the signal. The low signal level will also cause the audio signal-to-noise ratio to be low, regardless of how well the tuner has been designed to do Job Two, because of the theoretical limits of FM reception. The job of the tuner under these conditions is to reproduce the signal with a performance level as close to the theoretical limits as possible.

As you would expect, some tuners do one job better than the other. For this reason, as we shall see, there is no "best" tuner. Further, it is important to understand that the tuner's specifications can tell you a lot about how well it will do the job of receiving the signal, but they are by no means complete. The signal conditions that are present at the input of a tuner are very complex, and there is no substitute for connecting the tuner into your signal environment to see how well it performs. (Just to make sure even the far-gone tweaks understand this, we are talking about noise and distortion levels here, not "slam" or "pace" or some other tweako thing.)

So why do you need to read this article? Well, if you have the need for a super tuner to do Job One well, the best thing would be to convince all the dealers in town to loan you all their best tuners. You could then pick the one that gave the best results at your home location. Unfortunately, we have not encountered such friendly dealers. So we'll try to give you some signposts regarding the small group of tuners that may be best for your signal conditions. That way you'll only have to try a couple of tuners at home. But please do not run down to the store on just our advice and purchase something you cannot return because what worked great for us may not work great for you.

You should also note that tuners have dozens of internal adjustments. If they are not set correctly, then all bets are off. We have encountered many tuners that were not adjusted properly. No doubt you may also end up with a misadjusted sample. Thus you might have the perfect tuner and never know it because it is not performing properly. Getting a tuner properly adjusted is not an easy job. Proper adjustment of tuners often requires equipment that is not in your average TV repair shop, such as a low-distortion stereo FM signal generator and a high-precision distortion analyzer. In addition, the adjustments often interact, and thus the process of adjusting a tuner properly requires a large amount of time. If you have a dealer who has the equipment to do the job (ask to see it!), you are better off purchasing the unit from him than saving 5% on mail order.

It is a lot easier to assess how well a tuner will do Job Two. As we shall see below, many manufacturers make big cost-cutting moves that are easily identifiable and measurable.

That said, we should also point out that designing a tuner is a much bigger problem than designing an amp or preamp. Most designers of tweako audio stuff would not know where to start.

The ins and outs of demodulation.

I am going to reverse things and look at Job Two before Job One. Before we discuss how a signal is decoded by the tuner, we need to know how the signal is encoded (modulated). Frequency Modulation involves assigning to the transmitted signal an instantaneous frequency (for techie folks this is the derivative of the signal's time-varying phase) that represents the amplitude of the information signal. As the amplitude of the incoming signal changes so does the instantaneous frequency. A voltage-controlled oscillator, as the name indicates, is a device that will do frequency modulation, since a voltage controls the frequency of oscillation. Now, the VCO must be very low-noise or the noise will also be transmitted. In addition, the VCO must be very linear in mapping the signal's amplitude to the instantaneous frequency. Any nonlinearity will result in distortion. You have no control of this, since the VCO is on the site of the FM transmitter. So, if the station you want to receive has poor equipment, it does not matter how good your tuner is.

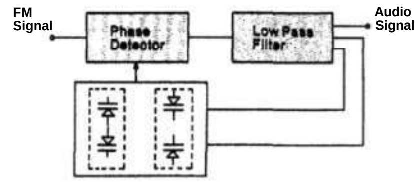
Now, it should be clear that the louder the signal gets, the further the frequency deviates from the zero-input-signal frequency (the carrier). The FCC restricts how far a signal can deviate, lest a signal from one station should wind up in the space of another. The FCC thus sets the maximum frequency deviation from the carrier frequency that a transmitted signal can have (75 kHz for FM). Since the process of Frequency Modulation is nonlinear, the spectrum of the signal at the output of the VCO can be much more than twice the frequency deviation. To understand this we need to use the Bessel function [Cook 1968].

[This section contains too much math and has been censored by the Editor.—DAR]

We can thus see that the typical FM spectrum occupies from 150 to 200 kHz. To prevent exceeding the FCC limits, stations put limiters in the audio signal path. Because loud stations attract more listeners, many stations compress the audio signal and then let the limiter come on often. Nothing can be done to fix the signal once it is mangled in this manner. Classical, jazz, and some NPR stations do try to produce a low-distortion, low-noise signal with a wide dynamic range, worthy of the tuners covered in this article. But it must be noted that even these stations will use some compression to prevent the signal from becoming inaudible on cheap equipment or in a car.

So how should we demodulate an FM signal? The optimal receiver in communication systems often involves placing the encoder in a feedback loop. For the case of FM, we would have the decoder's output connected to the input of a VCO (the modulator for FM signals). We then need a mechanism to compare the output of the VCO and the incoming signal, so that the VCO's instantaneous frequency matches the instantaneous frequency of the incoming signal. An appropriate mechanism is a phase detector which compares the phases of two signals. (Recall that the instantaneous frequency of a signal is related to the phase of the signal.) The error signal at the output of the phase detector is filtered to stabilize the loop and returned to the VCO input to close the loop. The circuit that has just been described is a phase-looked loop (Figure 1). It has been shown that the PLL is indeed close to the optimum demodulator for FM [Viterbi 1966]. As we shall see, many high-end FM tuners use a PLL demodulator. This a relatively recent development because, as stated above, the VCO must be very linear and have very low noise. Only recently have the Japanese designers been able to make such a high-performance VCO available at low cost. An interesting property of a PLL is that it can demodulate a signal with a lower signal-to-noise ratio at the antenna terminal than other types of demodulators [Panter 1965].

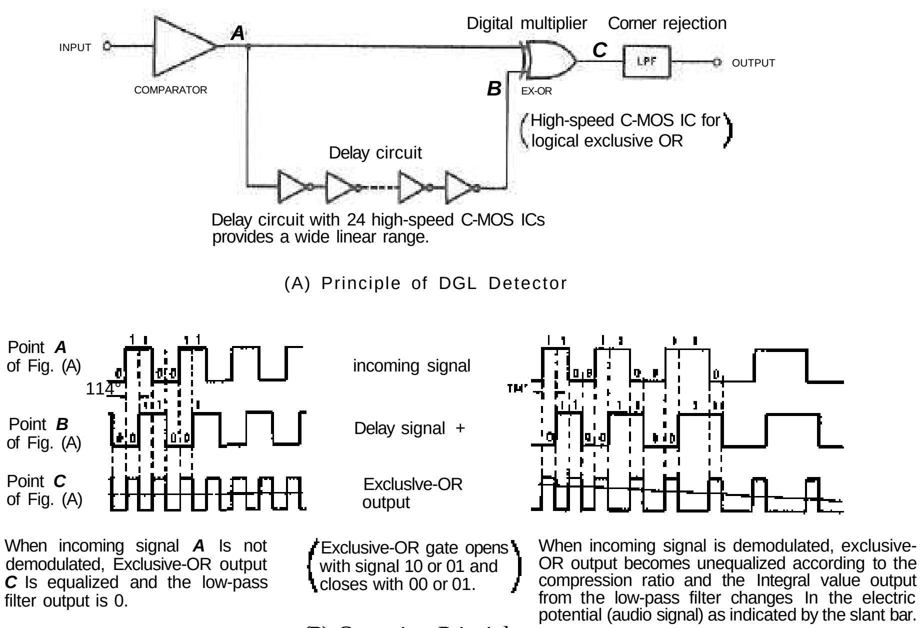
Figure 1: Phase-locked loop FM demodulator with high-performance VCO.



Dynamic Twin Drive V.C.0

One advantage of a PLL is that the bandwidth of the demodulator can be easily changed. For the most faithful demodulation of the signal, the loop should have a wide bandwidth. This is the condition we want for a good, clean signal, but if the incoming signal is noisy, then we want to reduce the loop bandwidth to reduce the effect of the noise [Gardner 1979]. Another key advantage of the PLL is that the output signal level is independent of the amplitude of the incoming signal (the reduced signal level may reduce loop bandwidth, however; see [Gardner 1979]). Since noise and interfering signals will cause the incoming signal to have significant amplitude variation, we want the detector to ignore these variations. The specification that tells us how well a tuner rejects AM signals is the AM rejection. AM suppression is defined in terms of the relative disturbance caused by amplitude modulation when the carrier is simultaneously amplitude- and frequency-modulated [IEEE 1975]. This is a key specification to determine how well the tuner rejects interference such as fading, multipath, airplane flutter, lightning, electrical equipment noise, etc. Super tuners should be in the 80 dB range. Care must be taken with this specification because it gets better at higher RF signal levels, and most data sheets do not give the level at which the test was performed.

The pulse-count demodulator is another high-performance circuit (Figure 2 shows an implementation by Accuphase). Based on theoretical analysis this circuit



(B) Operation Principle

Figure 2:
Pulse-count detector (or demodulator) in an advanced implementation by Accuphase.

should not be as good as the PLL but in practice it is much simpler to implement, and thus the practical results may meet or exceed PLL performance. It is inherently a very linear and very low-noise design. It is also easy to see how it works. Every time the incoming signal crosses from a positive value to a negative value (a zero crossing), the circuit generates a pulse of fixed duration. (In the Accuphase implementation shown in Figure 2, a zero crossing from negative to positive also generates a pulse.) If the instantaneous frequency is high, the pulses bunch together. If the instantaneous frequency is low, then the pulses are farther apart. Filtering the pulse stream with a lowpass filter averages the pulse count over time, and this yields the demodulated FM signal. This works because closely spaced pulses have a high average value and pulses farther apart have a lower average value. Like the PLL, the pulse-count detectors have very good AM suppression because only the zero crossings of the incoming signals are used. The problem with this type of detector is that it requires a double-conversion IF stage that reduces the IF frequency from the typical 10.7 MHz to a lower frequency in the 2 MHz range. This adds significant complexity, although it has an additional advantage of improving selectivity in some cases, since it may be easier to build a selective bandpass filter if it has a lower center frequency [Carson 1990]. Double conversion is required so that the frequency deviation of the signal is a significant percentage of the carrier frequency. The higher this ratio, the lower the output of the detector. Note that you cannot directly convert to a 2 MHz IF because the image rejection of the tuner would become very poor (see below).

We have one more detector to look at (we are leaving out a bunch of older or less popular types). This detector is significant not because it is very good but because it is very common, being (yes, you guessed it) cheap to make. The detector is the quadrature detector. In the quadrature detector a two-step approach is used. First the signal is differentiated, then it is sent to an AM demodulator. Both stages of the process are difficult to realize with precision. In the quadrature detector a simple tuned circuit near resonance provides a nearly constant phase characteristic that is used to approximate a delay line. This delay fine is then used to approximate a differentiator. Thus we have two levels of approximation to the differentiator [Clark 1971]. A balanced mixer is then used as a synchronous detector for the AM signal that comes from the differentiator.

What is a balanced mixer? It is a simplified form of a circuit called an analog multiplier. A true analog multiplier takes in two analog signals and multiplies them together. It can work in the voltage, current, or charge domain. Typically it is voltage-in-voltage-out, so if port A was at 2 volts and port B at -3 volts, the output of the analog multiplier would be -6 volts. The term *balanced* is used to indicate that none of the signals at the inputs of the mixer will appear directly at the output as distortion components. Harmonics of the multiplied signals may appear in a balanced mixer, and this differs from a true analog multiplier. These harmonics result because one input port of the mixer may significantly distort the incoming signal. Even-order harmonic distortion components from this port do not become involved in the mixing process in a balanced mixer, but the odd ones will. In an unbalanced mixer all harmonics are involved. The balanced mixer is easy to integrate, so the only external component is the tuned circuit. That is what makes it cheap. Unfortunately, given the two layers of approximation, distortion performance is never as good as with the PLL or pulse-count detector. Worse, the tuned circuit must be tuned to give the correct delay or the distortion is even higher (see the Denon review below). As many as three different adjustments may be on the tuned circuit, and these adjustments require the equipment your TV repairman does not have. If the frequency deviation becomes large, the approximations break down completely and significant distortion can result. Signals with interference on them require a detector with a wide bandwidth if the interfering signals are to be rejected [Panter 1965], so the limited performance of the quadrature detector presents a problem here. In addition, the quadrature detector cannot reject AM on its input, so circuits preceding it must limit the signal to provide any rejection of AM. Signals with large levels of interference will have a large AM component, so we are in trouble again.

Stereo FM explained.

Once we have demodulated the FM signal, we must now recover the two stereo channels. Several methods are used to modulate the stereo signal. I will describe the one that I think is easier to understand. A switch alternately samples the left and right channels at a 38 kHz rate. Of course, the signals must be bandlimited before sampling, just as in digital audio. The band limit is 15 kHz. Now, a mono receiver will reproduce both channels, since it will just average the alternate samples together. In a stereo receiver another switch is synchronized with the original switch, so when the switch at the transmitter is connected to the left channel the switch in the receiver will also be connected to the left channel, Then the switches at both the transmitter and the receiver go to the right channel (Figure 3). A filter follows the switches to remove out-of-band signals, again for the same reason this is done in digital audio. Often it's a pretty sloppy filter, so tuners often have frequency-response specs that allow a loss of 1 dB at 15 kHz. The "subcarrier product rejection" is the specification that indicates how well the filter is rejecting these out-of-band signals.

So far so simple, but how do we synchronize the switches? What is done at the transmitter is to send a 19 kHz pilot tone that is one half the switching frequency. At the receiver a PLL (this is another PLL, not to be confused with the PLL detector above) locks onto this signal. We want the output of the VCO to run at twice the speed of the 19 kHz pilot, since that is the switching frequency required. A digital divide-by-2 is placed between the VCO and the phase detector, so the VCO runs at 38 kHz instead of 19 kHz. The 38 kHz signal from the PLL drives the switches. Note that a simple VCO that generates a square wave can be used to drive the switches. Also, this VCO does not have the linearity requirements of an FM detector because it has to run at just one frequency.

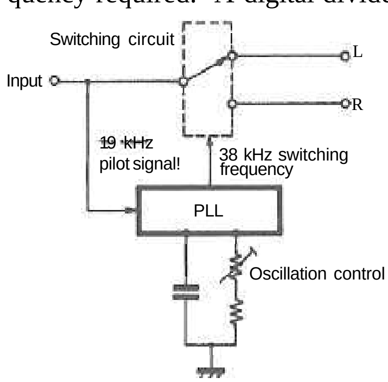


Figure 3:
Basic configuration of a stereo demodulator.

Often the VCO operates at a much higher frequency, with larger divider chains separating it from the switches. This allows the production of a signal with less phase noise. *(Jitter* is the term we would use if were examining the VCO's output in the time domain. Time-domain jitter and phase noise are mathematically related, as discussed in Bob Adams's article in Issue No. 21.) Phase noise on the pilot signal will lead to increased noise and distortion at the audio outputs, as well as reduced channel separation. As an example of very high-frequency VCO operation, most Sanyo chips run at 456 kHz. That allows them to use an external mechanical resonator, which will have even lower phase noise. This is a high-Q mechanical element similar to a quartz crystal but made of different material; it is usually used at frequencies where quartz crystals are uneconomic. Also, the resonator is superior to an RC-based VCO in three important areas: it is more accurate and does not need adjustment; it is not sensitive to temperature changes; and it has excellent long-term stability. Because of this the lock range of the PLL can be greatly reduced. That allows the loop bandwidth to be narrowed, making the PLL less sensitive to noise on the incoming signal. A design by Delco Electronics [Manlove 1992] uses a different architecture to accomplish a similar thing. The centerpiece of the Delco system is an analog/digital PLL that is too complex to discuss here. Delco reports they reduced the lock range of the PLL from a typical value of 800 Hz down 2 Hz.

Of course, some of us can hear the 19 kHz pilot, so it has to be removed. The el cheapo approach is to use a lowpass or notch filter at the output of the demultiplexer. Thus we have more stuff in the signal path to create frequency-response errors and add distortion. The better approach is to remove it by cancellation. Since we have made a replica of it with the PLL, this is relatively easy, but first the square-wave signal from the VCO must be filtered, since the transmitted pilot tone is a sine wave and the replica signal is a square wave. Then the signal level must be trimmed, using a pot to exactly cancel the pilot tone. The Delco folks have a clever way of doing this without external components to filter the signal or trim the amplitude or phase of the summer [Manlove 1992], but it is too complex to go into here.

We have a big problem with the multiplex decoder outlined above. The switches that are selecting the left or right channel will also demodulate any signal that corresponds to a harmonic of 38 kHz. Such signals will be present as a result of spurious frequencies created by adjacent and alternate channels and by RF intermodulation. Adding to the fun is the fact that subcarriers containing Muzak or digital data such as paging signals are also sent along with the FM signal you are listening to. *[The politically corrupt and cynically compromised FM stereo standard bulled through the FCC bureaucracy in 1961 is the reason for this.—Ed.]* These adjacent-channel frequencies, when demodulated, cause beat interference called "birdies." (If you listen to a weak stereo signal on your tuner during a quiet passage you will know why they call it birdies.) Removing the interfering signals with a sharp filter before the switches will clean up the birdies (they call it an antibirdie filter) but will also cause phase shifts that prevent synchronization of the switches to the transmitted left and right channels. That results in distortion and reduced channel separation. A specification called "SCA rejection" indicates how quiet the tuner is when the Muzak subcarrier is present.

A better approach is to use a sinusoidal 38 kHz signal and an analog multiplier, instead of the square-wave-driven switch. No demodulation of out-of-band signals can occur in such an arrangement. The problem is that you need to have a high-performance analog multiplier. These are open-loop devices; they have no feedback loop in the multiplier core to improve the accuracy of multiplication. Multiplication accuracy to 0.1% is thus very difficult to achieve. Obviously, any inaccuracy in the multiplication will give rise to distortion. It is difficult to design analog multipliers to have good noise performance. The Rotel RHT-10 tuner reviewed below is the first tuner to my knowledge to do this. (Something did not turn out quite right with the design because an antibirdie filter is still in the signal path.)

Pioneer also uses a sinusoidal 38 kHz signal. They combined this with a pulse-count detector (see explanation above), since the output of the pulse-count detector is binary. They connect the input of the switch to the 38 kHz signal and switch it with the binary signal from the pulse-count detector [Ishida 1984]. The problem with this approach is that it only works with a pulse-count demodulator. Pioneer appears to have extended the approach to the use of PLL and quadrature detectors in their current product, but how it works using these demodulators has not been disclosed. (I tried to get an answer from the engineers at Pioneer, but their response was to refer me back to the original patent they had filed, which discusses only the use of pulse-count detectors. Either my question did not get translated into Japanese correctly, or the engineers just do not want to disclose what they are up to.)

Sansui uses another approach that allows for full integration of the demultiplexer (another term commonly used for the stereo demodulator or stereo decoder). Sansui attempts to approximate a sine wave using Walsh functions [McGillem 1974]. The Walsh functions are a set of periodic rectangular pulses which, when combined at different amplitudes, can approximate an arbitrary periodic waveform. The Fourier series does the same thing but with a set of periodic sine waves. What makes the Walsh functions ideal for integrated systems is that summed scaled rectangular pulses are easy to generate on an IC. It turns out that two scaled Walsh functions are all that is needed to eliminate all the harmonics in the PLL output, up to the 8th harmonic [Takahashi 1985]. To generate the correct Walsh waveforms, the VCO in the Sansui chip runs at 304 kHz. Since Sansui has fallen on hard times, you cannot purchase this interesting tuner (TU-X701) with the above chip and a nice PLL FM detector anymore.

Chips by Allegro, Sanyo, and Sony use a similar scheme, although they have not published the detail that Sansui has. Antibirdie filters are typically not used in front of these chips. The team from Delco Electronics [Manlove 1992] also uses a similar idea, although the complete circuit implementation is significantly different. Delco uses more periodic rectangular functions to get the job done. This removes even more harmonics.

You may have noticed that this is the third time I am mentioning this Delco chip, so you might want one in your tuner. Unfortunately, to my knowledge, Delco does not sell chips, so this chip can be found only in GM car radios. Yes, the FM demultiplexer chip in your car is quite possibly better than the chip in your tuner.

Noise in an FM system rises with the audio frequency [Taub 1971]. For this reason the signal transmitted at the FM station is pre-emphasized, so that the signal rises 6 dB per octave above 2 kHz. A matching de-emphasis circuit is in the receiver. This de-emphasis circuit counteracts the rise in the noise. The problem with this scheme is that the signal levels at frequencies above 2 kHz must remain low, or else the station will overmodulate. Acoustic music and speech luckily have spectral densities that decrease with frequency above 2 kHz.

In a stereo transmission, the mono signal occupies the same spectral band as it would if a mono-only transmission were taking place (see explanation above). That was an FCC requirement in order to insure that a mono receiver could receive a stereo signal. The information about the stereo signal is contained higher up in the spectrum (Figure 4). These new sidebands come from the sampling action of the switches at the transmitter, which run at 38 kHz. It can be shown (but not here, says the Ed.) that these sidebands contain the signal L-R. When you add L-R to the mono signal, you get the left channel. Subtract it and you get the right channel. It can be shown that the sampling action of the switching circuit in the

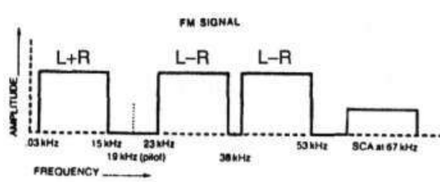


Figure 4: Demodulated spectrum of a stereo FM signal.

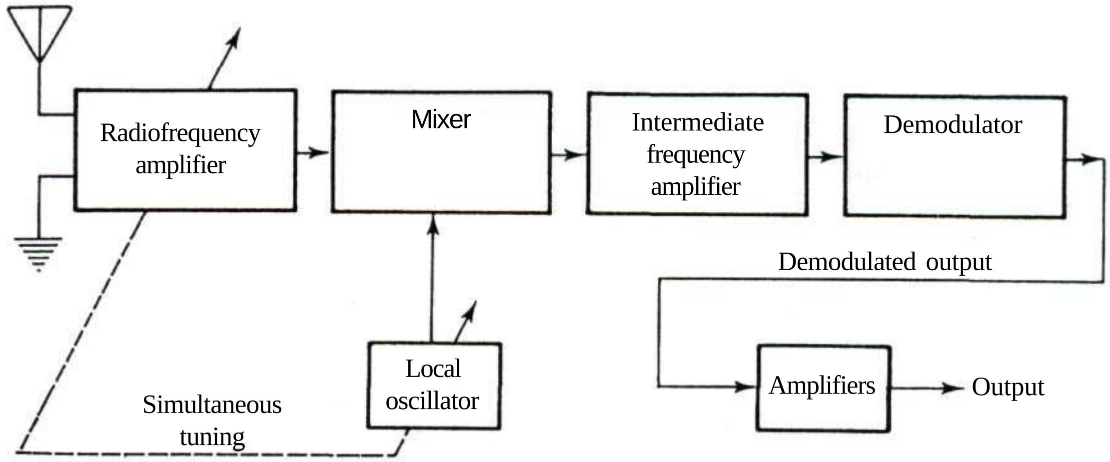


Figure 5: Simplified block diagram of a superheterodyne receiver.

multiplex demodulator does this function, but do not worry if this is not clear. What you need to know is that a mono FM signal has a bandwidth to 15 kHz, but a stereo FM signal goes out to 53 kHz. Now recall that noise increases with frequency in FM. So the L-R information in the 23 to 53 kHz portion of the composite signal has a lot more noise. How much poorer is the signal-to noise-ratio of the stereo information? An order of magnitude poorer, 22 dB worse to be exact [Taub 1971].

If the incoming signal is strong and clean, the theoretical signal-to-noise ratio is very high and this 22 dB performance deficit is not significant, since other noise sources dominate. The signal-to-noise ratio then should, ideally, become the value produced by the tuner when it is receiving a mono signal. If the signal is weak, then we can hear every bit of the 22 dB noise penalty when we switch from stereo to mono. It turns out that the ear tolerates fairly low channel separation at higher frequencies, so a blend circuit is often used to reduce the channel separation (and hence the noise) in the upper frequencies. Pioneer takes this one step further. They have a set of blend circuits, each in a specific passband. A circuit looks at the noise in each passband and adjusts the amount of blend up or down in response to the noise in that band. This dynamic adjustment gives a better stereo image, with less noise on weak signals. The downside of this system is that it is very complex.

Carver (the man and the company) varies the amount of L-R signal used, depending on how different in level and content L is from R [Feldman 1982]. Carver creates two signals, L/R and (L+R)/(L-R), and uses them to sense how different the levels of L and R are. Carver also uses more L-R when the leading edge (fast, short-term information) of a signal is detected. Carver says he uses a psychoacoustic phenomenon known as the precedence effect. When leading-edge information occurs, it is critical to the localization process. At other times, Carver uses a phony L-R signal concocted from the low-noise L+R. He does this with what he calls a phase randomizer and a spectral shaping circuit. It would appear that all this should not sound at all like the original stereo signal, but in practice it works very well, at least to my ears. Is the Pioneer approach better than the Carver approach? Stay tuned (no pun intended); we intend to tell you when we have both tuners in our laboratory back to back.

Pioneer has one more trick up their sleeve. They noticed that most of the noise and interference occurs above 38 kHz. They also determined that they only need the information from 23 kHz to 38 kHz (for the techie crowd, this is the lower sideband of the L-R as seen in Figure 4) because the upper and lower sidebands contain the same information. Now Pioneer uses only the information in this band (for the techie folks, they use a single-sideband demodulator). Unfortunately, this process introduces some distortion in actual practice, and the THD of the decoder is in the 2% range. That is very listenable, however, in the context of a weak FM radio signal.

The front-end circuitry.

OK, it is now time to look at the front end of the tuner, since we started in the middle of the signal path. The job of the front end is to provide the *sensitivity* to detect the presence of the desired signal and the *selectivity* to accept the selected station and reject all other stations. Figure 5 shows a superheterodyne receiver. The RF amplifier, the local oscillator, and the mixer are called the front end. In a superheterodyne receiver the incoming RF signal is mixed with a signal from the local oscillator, changing for each station, in such a way that the output has the same center frequency for all stations. This approach allows for excellent selectivity because the tuned circuits in the IF strip that reject interference are *fixed* tuned circuits. This results in much sharper cutoffs than if the filters had to be variable, as they do in the RF stages. In addition, it is possible to have higher-gain amplifiers at lower frequencies.

The RF stage amplifies the weak signals to a level at which the mixer can work properly; thus it increases sensitivity. If the signal were strong you would think you could bypass the RF stage, but you cannot for two important reasons. The first is local oscillator radiation. If the RF stage did not exist, the signals from the local oscillator could get back into the antenna through capacitive and inductive coupling. Since the local oscillator runs between 98.6 MHz and 118.6 MHz, this is a good way to cause interference right in the FM band.

The second thing the RF stage does is improving image rejection. In a superheterodyne receiver, a problem at the mixer occurs because the mixer can translate signals both above and below the local oscillator frequency, and it thus is possible for an undesired station to get translated to the IF frequency. In most FM receivers, the local oscillator is set above the incoming RF signal; thus the difference frequency contains the desired signal at the output of the mixer. The image frequency is above the local oscillator frequency. The image frequency is thus the desired RF signal's frequency plus twice the IF frequency. Now, if the IF frequency is 10.7 MHz, the image frequency can never be in a broadcast FM signal. That is not an accident; we do not want an FM broadcast signal to become an image signal. The FM band is 20 MHz wide (87.9 MHz to 107.9 MHz), and since the image signal is at the RF signal frequency plus *twice* the IF frequency, any IF frequency greater than half the width of the FM band will work. They put the IF frequency as close as possible to this limit, because the lower the IF frequency the easier it is to build a selective filter and create high-gain amplifiers. (For homework show why the IF frequency of an AM receiver is 455 kHz). So we are concerned with what is on the air between 109.3 MHz and 129.3MHz. What is there usually consists of not very large signals because these are frequencies for aviation activities. Note that if the local oscillator had been chosen to be below the RF frequency, the images—in this case the incoming signal frequency minus 21.4 Mhz—would come from TV channels 3 to 6, at 60 MHz to 88 MHz, each TV channel being 6 MHz wide [Cook 1968]. So now you know why that is not done. The specification for a broadcast tuner that tells you how well it rejects images is called the "image-response ratio" or "image-rejection ratio" [IEEE 1975].

Unfortunately, other image interference is possible, because the mixing signal often contains harmonics. These harmonics also generate sum and difference products that can move undesired signals to the IF strip. TV channels 9 to 13 (186 MHz to 216 MHz) are sources from which second-order images, caused by the second harmonics of the oscillator, can get into the IF strip. Filtering the RF signal beforehand and attenuating the undesired one will prevent these images from contaminating the desired signal. Third- or higher-order images are rarely a source of interference because they are very highly attenuated by the tuned RF amplifier [Cook 1968].

Another interference is caused by spurious response. Spurious response is caused by nonlinearity of the RF amplifier or mixer. This can cause intermodulation products which again can fall into the IF band. Under worst-case conditions, with a heavily overloaded RF front end, a station may appear at a number of places on the dial. A common mechanism producing a spurious signal is the second harmonic of the local oscillator beating against the second harmonic of an RF signal. The IEEE test for this [IEEE 1975] is called the "characteristic-frequency test." In this test the incoming frequency is set to 103.35 MHz and the receiver is tuned to 98 MHz. The tuned circuit before the first RF stage helps here also by filtering out undesired signals before they get to the tuner.

Other scenarios involve two incoming signals and the local oscillator together forming a spurious signal [Cook 1968]. The IEEE standard test for RF intermodulation involves testing for the condition where the second harmonic of one interfering signal at 98.8 MHz is mixed with another interfering signal at 99.6 MHz. The mixed signal is at 98 MHz. The test is reported as the "two-signal spurious-response ratio." The frequencies of the two signals are too close together for the RF stage to provide any filtering. The test thus checks for how well the RF and mixer stages are designed by checking to see if they generate intermodulation distortion.

The IEEE standard allows the worst-case result of the tests for characteristic frequency and two-signal spurious response ratio to be reported as a single number called the "spurious response ratio," representing the poorest of these measurements. Unfortunately, this is often not followed in specification sheets, and it appears that the characteristic-frequency spurious response ratio is then published as the spurious response ratio, period.

In the case of a very strong station, it is important that some method of signal attenuation be supplied at the

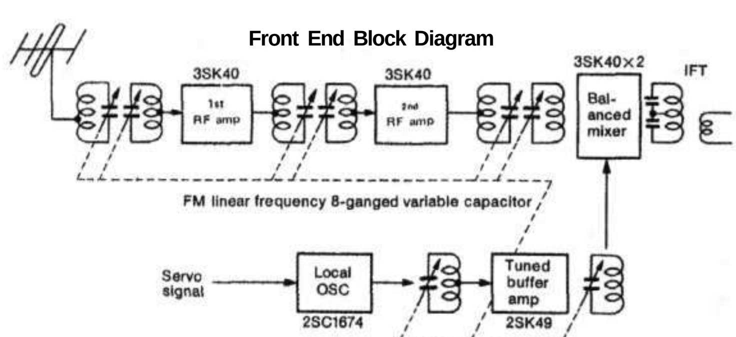


Figure 6: RF stage of the now defunct Technics "Professional Series" ST-9030 FM stereo tuner (from the good old days when front ends were front ends).

input of the RF stage to prevent overload. In some tuners this is automatically controlled by an automatic gain circuit. Other tuners have manual controls that enable the attenuation. The manual controls are nice because sometimes the AGC gets things wrong. Consider the case of a relatively weak-signal channel adjacent to a very strong signal. The AGC would turn off the attenuation to gain up the signal, but in doing so it would cause the strong adjacent-channel signal to overload the RF stage. Smart AGC circuits, such as used in the National Semiconductor LM1865, can be designed to look for strong adjacent-channel interferennce before increasing RF gain.

In the good old days, super tuners had super front ends, For example, the Technics ST-9030 had an eight-gang tuning capacitor and two RF amplifier stages (Figure 6). The ST-9030 had an image and spurious-response rejection of 135 dB because of this complex RF stage. The eight gangs added up the following way: One was for the local oscillator, and one was for a filter after the local oscillator to remove harmonics. Each of the RF stages had a double-tuned filter preceding it, as did the mixer. Each double-tuned section needed two gangs. A double-tuned section provides a flatter passband and a steeper rolloff rate than a single-tuned section. Today you get one RF amplifier and the equivalent of three or five gangs. The result is that some of the best tuners today have characteristic-frequency spurious-response and image-rejection ratios of only 80 dB. In addition, they may lack sufficient RF gain, especially when operated from an indoor antenna. The Magnum Dynalab "Signal Sleuth" is an extra stage of tuned RF gain which can be added to any tuner. This device can solve some difficult signal problems that some of the modern super tuners cannot handle. The Signal Sleuth is reviewed in this issue.

Another important parameter that the RF amplifier should satisfy is constant input impedance, independent of frequency. The need for this is explained in the indoor antenna reviews found elsewhere in this issue. As in the case of antennas, the deviation from the ideal input impedance can be given as a voltage standing-wave ratio (VSWR). This is how Accuphase gives it (as usual they have the most complete set of specifications). Most manufactures do not give the specification at all.

In modern receivers, all the tuned circuits are tuned with voltage-controlled capacitors that use a diode assembly. This is called a varactor. The voltage range to tune the varactor should be large, so that the time-varying information signals present at one end of the varactor cannot significantly change its capacitance. Signal-dependent changes in capacitance can give rise to modulation components. Early varactors had this problem, but voltage swings for tuning over the FM band are now 20 V for modern varactors, and the information-signal swing is much smaller than this.

The voltage for the varactors comes from the loop filter that forms yet another PLL. The local oscillator is the VCO of this PLL. A set of digital dividers after the VCO is used to tune the tuner. The output of the dividers is connected to a phase detector. The other input of the phase detector comes from another digital divider that has as its input a crystal reference oscillator. Under microprocessor control, the dividers are set so that the local oscillator runs at the correct frequency to move the desired signal to 10.7 MHz. The voltage at the output of the PLL loop filter also adjusts the tuned circuit's RF stage and mixer to the correct position to receive the desired station. Clearly, no noise should be on the voltage that is connected to the varactor in the VCO because this will give rise to phase noise in the local oscillator. Since the PLL is a closed-loop system, the loop filter cannot be designed to provide an arbitrary amount of filtering because this will cause the PLL to go unstable [Gardner 1979]. The best way to insure a quiet varactor input voltage is to run the phase comparator at as high a frequency as possible. The undesired signals at the output of the phase comparator can then be more easily removed by the loop filter because they are high in frequency.

Early digital tuners with slow IC technology had a problem running at high phase-comparator speeds and so they acquired a bad reputation. The phase comparators also had problems with dead bands that made things worse. (A dead band is a region where the phase detector gives no change in output even when the phase difference of the incoming signal changes.) Modern ICs run at higher frequencies and use high-performance phase comparators; thus modern tuners do not suffer a performance penalty when they use digital tuning. RF noise from the microprocessor and other digital circuits also caused noise problems in early frequency-synthesized tuners. These problems do not occur in modern designs because the problem is understood and techniques to minimize the interference are known. Linearity of the VCO is not a problem in digital tuning because only one frequency has to be synthesized to receive a given station.

The problem with digital tuning is that the station may not be at the exactly correct frequency. This is especially true of stations retranslated to other spots on the dial on FM cable systems. It is thus desirable that a fine-tuning mode be available to detune the receiver.

The mixer is a relatively simple circuit but very hard to optimize, according to Richard Modafferi, who has done state-of-the-art mixer designs. Obtaining sufficient dynamic range is the really tough design challenge. Two design parameters must be optimized to have good dynamic range: (1) noise levels in the mixer must be very low so that weak signals are received; (2) the mixer stage must be able to handle strong signals in a manner that does not allow intermodulation distortion to occur, which is a more difficult requirement here than in the RF stage because the mixer is not just an amplifier; it is also performing frequency translation. A lot of other design challenges also exist in mixer design, such as making sure the mixer does not load down the RF stage or interfere with the proper operation of the local oscillator. A balanced mixer will usually perform better than a single-ended one. In a balanced mixer, even-order harmonic distortion components from the local oscillator do not become involved in the mixing process. This reduces spurious interference. The downside of a balanced mixer is that it is more complex. Also, it must be designed very carefully if is to be truly balanced at RF frequencies.

The IF section amplifies the desired signal and removes the interfering signals before the FM detector. One thing the IF strip should *not* do is directly pick up any signals at its center frequency. This can happen because the IF strip has a lot of gain. 10.7 MHz happens to be kept relatively quiet by the FCC because of this problem. Correctly shielding the IF stage also helps. The IF rejection specification is a measure of how well the IF strip rejects this signal. In the IEEE test called the "IF response ratio" [IEEE 1975], an FM-modulated signal with a carrier at 10.7 MHz is applied to the antenna terminals.

The gain requirements for the IF strip are quite high. At the antenna input a signal may be as small as 1 μV. The FM detector may require a signal of 1 V. That adds up to 120 dB of gain. The RF section is going to have about 30 dB of gain, so the IF strip needs to have a gain of 90 dB. Fortunately the stage does not have to be linear, so feedback is not used and the gain is achieved by cascading individual stages of open-loop gain. This does not prevent the possibility of oscillation in the IF strip through parasitic coupling between the output and input of the strip. Because of all the gain in a tuner, the designer must always be on guard against such problems.

In between each stage is a tuned filter. This is important because, as we gain up the desired signal, we also gain up the undesired signals. The filters reduce the amplitude of the interference before it is sent on to the next stage of gain. In this manner the desired signal amplitude becomes larger than the interfering signal.

For the FM signal to pass through the IF section without picking up distortion, it is important that none of the sideband components of the FM signal be shifted in phase relative to one other. (No, all you tweaks, this has nothing to do with the ear's sensitivity to phase. In FM we have modulated the signal so that the instantaneous frequency, and thus the phase, contain the information about the baseband signal. That is why phase integrity is important in FM signals.) Now we have a problem, since a sharp IF filter will reject the interfering signals, but it is also going to affect the phase. This is because you cannot have a sharp transition band in a filter without significant phase delay [McGillem 1974]. Consequently, a good tuner offers selectable IF filters. Wide filters give low distortion in the demodulated audio signal. Narrow filters give good rejection of interfering signals but have poorer audio performance. Some tuners even have an extra narrow mode for bringing in very difficult signals, at the penalty of even more audio signal distortion.

Modern filters in a tuner are mechanical resonant elements whose input is driven by, and whose output is picked up by, an electromechanical transducer, such as a piezoelectric element. The quartz crystal is a simple mechanical filter. In the IF strip multipole elements, often called ceramic filters, are used to give flat passbands and sharp rolloffs. These filters perform much better than the old double-tuned circuits they replace. Another advantage is that these filters do not have to be tuned. In the wide mode two filters may be in the signal path; in the narrow mode four filters. In extra narrow we can see five or more filters, and the passband of each of the filters will be decreased at the cost of phase distortion. The Pioneer Elite F-93 wins the contest of the most mechanical filters in the IF, with eight filters. The old Mcintosh MR-78 had a filter that placed transmission zeros at the alternate- and adjacent-signal positions. That appears to be a good idea. I cannot figure out why it is not used in other tuners. (The high cost of the filters is the likely reason.)

Pioneer came up with a very innovative solution to the "narrow filters distort the signal" problem. They used this solution in the F-91 tuner. It is such a good idea that it could have gotten someone a Ph.D. (Indeed, a related concept did get someone a Ph.D.—see [Rich 1991].) What the engineers at Pioneer observed was that, although the spectrum of an FM signal is wide when *averaged over time*, at any instant in time the signal has just one spectral line at the instantaneous frequency. So, if you build a *time-varying* filter with a very narrow bandwidth that moves with the instantaneous frequency of the desired signal, then you can eat your cake and have it too. The very narrow bandwidth removes the interferers, but the time-varying nature of the filter insures that the FM signal is not distorted. To move the filter around, a complete auxiliary IF strip designed in the conventional manner and a PLL FM detector are required. We now have a chicken and egg problem, since the conventional receiver has to place the passband of the time-varying filter in the correct place. This is what limits the performance of the system.

One very interesting (at least to this author) variation on this is to replace the bandpass filter with a notch filter. Then the stronger signal is removed, but a weaker cochannel interfering signal may now be revealed. Further information on this is in—you guessed it—[Rich 1991].

Surprisingly, the F-91's tracking filter was dropped from the newer Pioneer Elite F-93. The reason given by Pioneer was that the circuit needed very high-precision (read high-cost) parts and that performance nearly as good was achievable without them in the revised tuner. The F-93 reduces the distortion caused by the fixed narrowband IF filter by compensating for it in the FM demodulator.

Sony uses a variation on the tracking-filter theme that involves dynamically varying the position of the RF and mixer filters in response to the signal level at the output of the FM demodulator (which represents how far the instantaneous frequency has moved from the carrier frequency). This is easier to implement in the RF and mixer stages than in the IF because these filters are already tunable. One would think the large signal delay in the IF strip and detector would prevent this from working well, but Sony is shipping tuners with this thing, so it must do something positive.

One wonderful property of FM is that two cochannel signals can be very close in amplitude and yet only the stronger will emerge from the FM detector. This is the capture effect in FM systems. It does not exist in AM systems. A specification that tells you how much stronger a signal has to be to *capture* the receiver is the capture ratio. In a super tuner it is as small as 1 dB. The capture ratio is also important for a situation where multipath exists, since a multipath signal looks like a cochannel signal to the receiver [Panter 1965].

When multipath or a significant interferer is present, the amplitude of the IF signal can vary significantly [Panter 1965]. This does not affect the performance of the FM detector provided it has good AM rejection—strike one for the quadrature detector. When multipath or a significant interferer is present, the composite instantaneous frequency variation can be very wide; thus the FM detector must have a very wide linear range to accommodate these excursions—strike two for the quadrature detector. The widebanding works because the inband signal at the output of the FM demodulator turns out to be the desired signal. The interfering signal gives rise to distortion products that are not present inband. If the discriminator is not wideband enough, then inband distortion products may appear in addition to the out-of-band products [Panter 1965]. The bandwidth requirements rise steeply as the power ratio of the desired to the interfering signal closely approaches 1. Thus a detector which does not have wide bandwidth will have a poor capture ratio.

The quadrature detector does not strike out because of the limiter stage. The ideal limiter takes an input that is varying in both frequency and amplitude and provides an output of constant amplitude that retains the instantaneous frequency information of the transmitted signal [Cook 1968]. In essence it is a high-gain amplifier that is driven to clip. At some small signal level, the limiter will not have enough gain to clip, and the circuit will just amplify the signal and pass on the amplitude variations. Cascading multiple limiters reduces the signal level that is required to saturate the limiter. So the limiter removes the amplitude variations that the quadrature detector is sensitive to and gives the receiver its AM rejection, usually 50 to 60 dB. Of course, if the detector has some AM rejection, like the PLL or pulse-count detector, then the limiter is icing on the cake and the AM rejection goes to 80 dB.

The limiter can also help to reduce the bandwidth requirements at the detector in the presence of an interferer. This is done by the use of a cascaded narrowband limiter [Baghdady 1955]. What Baghdady found was that when the desired and interfering signal enter the limiter, the energy of the interfering signal spreads out in its spectrum. If you then send the signal through a narrowband filter, some of the interferer's energy will be removed. Continuing to do this a number of times will reduce the energy of the interfering signal relative to the stronger signal at the detector's input. This then eases the bandwidth requirements of the detector.

We have now arrived at the input of the FM detector, which is where we started at the beginning of this article. With a basic understanding of how an FM tuner operates, we can move on to the reviews. It should be noted that we did not hook all the fish in this first survey. Promising tuners not included are the **Pioneer Elite F-93** and **Sony ST-SA5ES** referred to above. Also promising are the very expensive **Accuphase T-109** that was also mentioned above and the bargain-priced **JVC FX1010TN** and **Yamaha TX-950.** Unfortunately, one of the great tuner manufactures of the past, Kenwood (great because they also design ham radio equipment and thus have a large and experienced design team) no longer brings any super tuners into this country.

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How I Evaluate FM Tuners

By Richard T. Modafferi
Technical Consultant to *The Audio Critic*

I've designed and built RF devices (transmitters, tuners, shortwave radios, cable TV amplifiers, etc.) for nearly 50 years, and thus one may assume in this regard that

(1) I know how to design and test RF devices, or

(2) I'm good at fooling people that I'm good at (1) above, or

(3) we're all ignorant and it doesn't matter.

Perhaps a combination of the three items above is closest to the truth. Regardless, the Editor has roped me into the job of testing, but not reviewing, tuners for *The Audio Critic*. An explanation of my test procedures is thus necessary.

An FM tuner is really just a radio. By this I mean that a tuner should work like a radio and receive as well as possible all signals present at its antenna terminals. Reception problems and the solution thereof are the largest challenge facing an FM tuner designer. Radio circuits in an FM tuner must work over a very large dynamic range, up to 120 dB in the best designs, as for example a lower limit of 1 μV and an upper limit of 1 V. In addition, FM tuner circuits have an astonishing amount of voltage gain, up to 130-140 dB in some designs. My MR-78, for example, has a voltage gain in its RF-IF circuits of about 137 dB! One μV at the 75 Ω antenna input produces 7 V at the input to the detector bridge, at an impedance of 52Ω. This is gain, folks. I'm not impressed by the bragging that goes on about about the dynamic range in the latest digital stuff. Tuner designers play with bigger numbers! The challenge of making tuner circuits work well over these big numbers involves a lot of study, experiment, and sleepless nights. The reward at the end, however, is a super tuner and good FM reception.

I share a mountaintop with two broadcast stations, an AM on 1430 kHz 600 feet away, and an FM on 92.1 MHz only 138 feet away. Big brother on 92.1 measures 1 V at my tuner antenna terminals. Another station on 105.7, across the valley, comes in at 0.25 V. Weak signals exist on 91.3, 91.7, 105.3, and 106.3 MHz. I try reception of the aforementioned weak signals in one of my tests for a tuner's reception capability.*

Tuner RF Performance Tests

(1) Outdoor tower-mounted high-gain antenna system. The tuner is connected to this antenna, and its ability to receive weak signals on 91.3, 91.7, 105.3, and 106.3 MHz is noted. The tuner must cope with a 1 V signal on 92.1 and a 0.25 V signal on 105.7.

(2) Selectivity test. I attempt reception of a weak NPR station on 91.3 (75 miles away), adjacent-channel to a local NPR station on 91.5 MHz (4 miles away). So far, I've found only three tuners that can receive 91.3, and these are the Onkyo T-9090II, my MR-78, and the Accuphase T-109.

(3) Indoor antenna reception test. The tuner is connected to a three-quarter wavelength "gamma-matched" vertical antenna, and reception quality on local stations is noted. Reception should be free of spurious responses. In some cases reception of fairly weak distant signals is possible.

(4) FM generator strong-signal test. The tuner is connected to an FM generator, and 1 kHz monophonic harmonic distortion is observed as signal is increased from 1 μV to 1 V. Distortion plus noise should drop from the initial value of 3-5% (-30.5 dB to -26 dB) at 1 μV to below 0.25% (-52 dB) at 10-15 μV and remain low up to the 1 V antenna input level.

(5) *$2f_1 \pm f_2$ test.* Two stations close in frequency will produce a pair of spurious signals whose frequencies are given by this formula. For example, 105.1 and 105.7 MHz combine to produce a "spurious" at 106.3. I look for it. This is the classic "stations coming in all over the dial" or "at wrong places on the dial" syndrome. The $2f_1 \pm f_2$ test is by far the toughest of the spurious-response tests. The front end in Figure 6 may do well on the other spurious-response tests and yet do very poorly on this one because the extra RF stage could cause intermodulation to get worse.

Baseband IM Distortion Test

The tuner is connected to a Sound Technology 1020A FM signal generator. A stereo signal of 1000 μV is applied to the left channel, modulated 100% at 10 kHz. Output from the right channel is observed. A spurious output tone of 1 kHz will appear, and in good tuners this tone should be 60 dB or more below 100% 1 kHz modulation (0 dB reference level). I devised this tough test during development of the MR-78 25 years ago. In 1968, only one tuner, the Marantz 10-B, could pass this test. The MR-78 passes. Today, almost any decent modern tuner passes also.

Residual-Junk Listening Test

This time I play music into the tuner from the Sound Technology 1020A. I modulate the left channel only and listen to the right. A perfect tuner would produce silence. Good tuners yield a clean low-level output. Even a small amount of distortion is audible because you listen to the "residual" channel, already down 40 to 50 dB because of separation. Here, 0.1 % (-60 dB) distortion, as referred to the other channel's output level, will be only 10 to 20 dB below the residual and hence clearly audible! Surprisingly, some tuners do well on this test, implying that their stereo decoders have very low distortion, and also that the tuner's entire circuitry has very good linearity.

This ends my regular test regime. I also perform some additional distortion and separation measurements, generally in order to verify the manufacturer's specs and to check tuner alignment. I do a touch-up alignment if needed.

***David Rich notes:*

Here in Audio Critic country, in Eastern Pennsylvania, reception problems abound. The principal classical-music station is 40 to 60 miles away (depending on the test site). Unfortunately, a 50,000-watt rock station is alternate channel to this station and about 5 to 15 miles away. This is the type of condition that separates the real tuners from the pretenders.

Over at the commercial-free low end of the dial, adjacent-channel stations are coming from all around the area. Some come from local 100-watt stations or 10-watt translators. The local NPR and college stations are quite clean but more often than not they play rock. Only when they play what a government-supported station should, not Classic B Sides, can we find out how well the tuners do Job Two. The marvelous Mercer County Classical Network comes in on a new translator. Before that it was 70 miles away and one of my principal worst-case test signals for this survey.

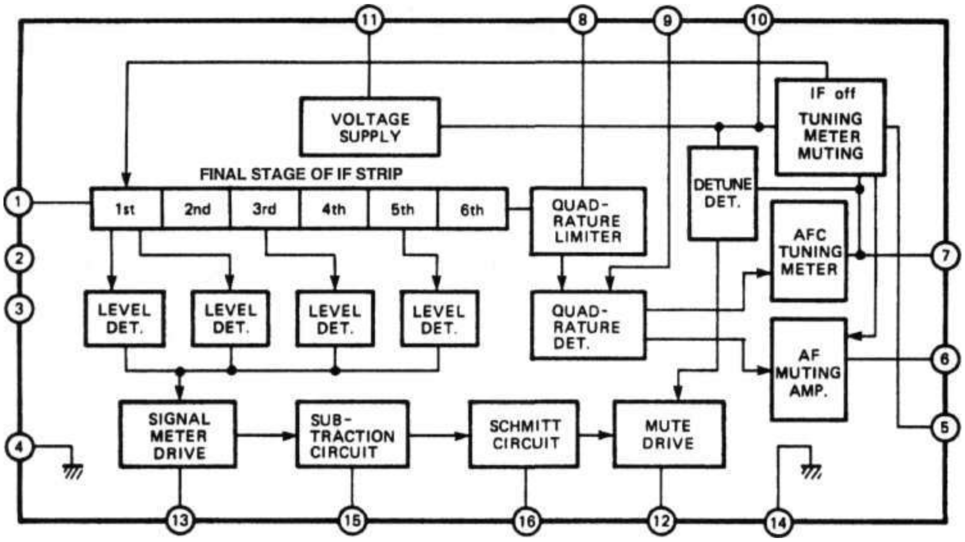

Figure 7: Block diagram of the ubiquitous Sanyo LA1235 chip (with the iniquitous quadrature detector).

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Denon TU-680NAB

Denon Electronics, a division of Denon Corporation (USA), 222 New Road, Parsippany, NJ 07054. Voice: (201) 575-7810. Fax: (201) 575-2532. TU-680NAB AM/FM stereo tuner with remote control, $600.00. Tested sample on loan from manufacturer.

The RF section of this tuner is reasonably well-done with the equivalent of five gangs, one at the RF input and two at the mixer section. The mixer is fully balanced—but to no avail, as we shall see below. Another pair of tuned stages is in the local oscillator. One is used to set the local oscillator's frequency; the other filters the signal from the local oscillator as part of a buffer stage.

The IF strip starts with a single discrete stage and uses two integrated amplifiers between the ceramic filters. Two filters are used in the wide mode. Four filters are used in the narrow mode, with an additional integrated IF amplifier also switched in when narrow mode is engaged.

The final stage of the IF strip and the FM demodulator are in the ubiquitous Sanyo LA1235. This chip is the most popular (it's real cheap!) IF detector. It uses a quadrature detector design, as shown in Figure 7. Now, Denon knows that a quadrature detector just won't cut it in a high-end tuner. They used an advanced PLL detector in the last generation of products that did not cost more. Two tank circuits in the quadrature detector must be adjusted for the tuning to be on channel center and for minimum distortion (try to get that one done by your local TV repairman), and the unit came to us incorrectly adjusted. Another adjustment involving a distortion measurement must be performed on the tank circuit connected between the mixer and the IF stage. The remaining adjustments should be within the scope of a competent repair person.

The multiplex decoder is another IC, the Allegro ULN3827. Allegro is a U.S. company that designed this chip primarily for car audio, and most of its specifications are not state-of-the-art for monolithic multiplexer chips. It uses a 608 kHz mechanical resonator in the VCO and has a 19 kHz pilot canceler that requires no adjustments. Walsh functions are used to improve rejection of spurious signals like the SCA carrier. Surprisingly, Denon still chooses to use an antibirdie filter. A dynamic blend circuit is included, which activates when the noise ratio becomes high. Most autoblend circuits (clearly a function you need more in a car than at home, where you can flick the blend switch) activate the blend on signal level, not noise level, and can blend a weak but clean signal accidentally. The PLL loop bandwidth in this chip is made wide for fast acquisition when no stereo signal is present and then is made narrow to increase noise immunity when the stereo signal is captured.

Another nice feature of the Denon is that the channel-separation adjustments are made individually for the narrow and wide modes to insure the best possible performance in the narrow mode. A pair of passive filters follows the multiplex decoder to remove the pilot tone and other higher-frequency products. The outputs are buffered with JRC NJM4558 devices.

The AM section is quite complex and performs much better than most AM tuners. It has a wider audio bandwidth and lower distortion than traditional AM tuners, as well as sophisticated noise-reduction circuits. It also has excellent sensitivity. (We can confirm that Rush Limbaugh sounded better on the Denon, but we performed no controlled listening tests.) If you have an AM station in your area that broadcasts something of interest, this may be the tuner for you.

A single 12 V supply powers the analog stages. Two 10kΩ resistors tied to the 12 V supply and a 100 µF filter capacitor form the analog ground path. You cannot get any cheaper than this. General construction quality is that of a mass-market product.

The FM RF front end has a very low noise figure. The 50 dB RF quieting occurred at 10.5 dBf. Most tuners are 20 dB noisier at this signal level. RF intermodulation performance was not so good, however. Strong stations prevented reception of nearby weaker stations that could be received by other tuners in this survey. This indicates that the RF stage (and/or mixer stage—remember that mixer stages are very hard to design) does not have adequate dynamic range, although the cause could not be identified by me from the schematic. Poor dynamic range can cause intermodulation of signals at the RF stage's input, since the stage becomes nonlinear.

Measured 1 kHz THD out of the box was -57dB for a 30,000 µV signal at 91.1 MHz. This fell short of the specified -60 dB. In addition, minimum distortion was not achieved when the unit was tuned dead on. The Modafferi 10 kHz stereo IM test gave a result of -56 dB on our test sample; on another sample tested by Modafferi the result was -69 dB. Clearly our sample was misaligned as delivered, and that includes the quadrature detector. Channel separation from 50 Hz to 15 kHz was >39 dB in wide mode and >33 dB in narrow mode. It never met the specified 50 dB at 1 kHz, measuring 47 dB in wide mode, but the full-band results are very good. Frequency response in stereo just made the +0.5 dB, -1.0 dB strip given in the manufacturer's specification sheet. That *could* be audible and should be tighter in a $600 tuner.

The Denon proved to be an average performer when presented with good signals and did poorly under difficult signal conditions. The AM performance is truly exemplary, and the circuitry to achieve this must have added to the cost of the unit. If AM performance is important to you, consider this unit. Others should pass it by, as I believe the less expensive Harman Kardon TU9600, JVC FX-1100BK, Sony ST-S550ES, and Yamaha TX-950 should at least match its FM performance at much less cost. The slightly more expensive the Rotel RT-990BX and Onkyo T-9090MKII will blow it away on FM. (They have no AM, so again if that is important, look into the Denon.)

Harman Kardon TU9600

Harman Kardon Incorporated, a Harman International Company, 80 Crossways Park West, Woodbury, NY 11797. Voice: (516) 496-3400. Fax: (516) 496-4868. TU-9600 "active tracking" AM/FM stereo tuner with remote control, $449.00. Tested sample on loan from manufacturer.

We have had this not very new but still current model in-house for some time, but for some mysterious "organizational" reason it never got to be measured by Rich Modafferi. *[Mea culpa.—Ed.]* I don't want to wait, however, to tell you what you can get for $449.

The front end and IF sections are pretty typical of units in this price range, but the FM demodulator and stereo decoder are anything but. The tuner has a state-of-the-art PLL FM demodulator, but to make it work at this price they could not design a high-linearity VCO. So they got real smart (yes, it is patented) and realized that the output of the VCO in a PLL must always track the incoming signal accurately, even if the transfer function from voltage to frequency of the VCO is not very linear. This is because of the phase detector used in the PLL.

What Harman Kardon does is to take the output of the VCO and send it through a Sanyo LA 1235 with its quadrature detector. No amplitude-modulation problem occurs in the quadrature detector because the VCO output is a constant and it's big. This combination does not allow the low-distortion properties of the PLL FM detector to be exploited but it does allow its excellent performance under poor signal conditions (including large amounts of AM) to be taken advantage of.

The loop bandwidth of the PLL is made narrow to exploit the circuit's special signal-demodulation properties under poor signal conditions; as stated in the main article, this may cause some distortion. The VCO output is thus an FM-modulated version of the desired signal cleaned up by the PLL. The quadrature detector then demodulates the cleaned-up signal.

OK, what more could you want for $449? Well, how about the top-of-the-line Sanyo multiplex decoder called the LA3450? It looks like a second source for the Sony CXA1064S chip used in the top-of-the-line Sony tuners. No other multiplex decoder has better specs than the Sanyo LA3450; it some cases it is significantly better. Yes, it has a pilot-tone canceler. Yes, the VCO runs at 456 kHz and requires no adjustment because it uses a mechanical resonator. Yes, it has no antibirdie filter because of a birdie noise-reduction system that appears to be similar to the Sansui approach. The decoded signal then goes, not to some cheap op-amp, but to a discrete amplifier designed in the Krell style. Other high-end touches can be found, although there are big limits at this price.

Measurements of the TU9600, plus more design details, will be published in the next issue.

Mcintosh MR7084

Mcintosh Laboratory, Inc., 2 Chambers Street, Binghamton, NY 13903-2699. Voice: (607) 723-3512. Fax: (607) 724-0549. MR7084 AM/FM stereo tuner, $1500.00. Tested sample on loan from manufacturer.

This somewhat baffling tuner was sent to us much later than the other units reviewed here and will therefore have to wait until the next issue for full coverage; meanwhile the Editor discusses it briefly elsewhere in this issue as part of the feature article on Mcintosh Laboratory.

Onkyo T-9090n

Onkyo USA Corporation, 200 Williams Drive, Ramsey, NJ 07446. Voice: (201) 825-7950. Fax: (201) 825-8150. T-9090II Quartz Synthesized FM Stereo Tuner, $790.00. Tested sample on loan from manufacturer.

This tuner has been with us for 11 years, with the Mark II revision appearing in 1988. When it first appeared it was the DX (distant reception) engine for the masses, with Richard Modafferi's Mcintosh MR-78 being the only thing that could outperform it. Today the T-9090II may be the ultimate super tuner for bad signal conditions, as the MR-78 is history and Mcintosh is unwilling to do an update on it. (Engineering has done a couple of designs but marketing has canceled full-scale development. The marketing folks say they would sell exactly six of them because the price would be in the middle five figures.) The only challenger in current production is the Accuphase T-109. On the difficult real-world signal tests presented to the tuner by Richard Modafferi, the Accuphase has slightly better spurious-response characteristics, but its selectivity in not quite as good. Since the Accuphase sells for almost four times the price of the Onkyo, we can definitively say that the Onkyo is the cost-effective DX engine.

The Onkyo has a number of nifty operational features. They include a variable gain stage that can even be operated by remote control. Two antenna inputs are switchable on the front panel. I have one on my FM cable; the other goes to the indoor antenna (cable FM does not carry many local stations in my area). Because crosstalk between the two antenna inputs is only 40 dB, this may not work in all cases if you are trying to receive a weak signal off the indoor antenna at a place on the dial where the cable company has placed a strong signal. Speaking of cable, the tuner shows center-tuned conditions with three LEDs. Since the tuning increment is 25 kHz, you can tune to the station's center even if it is not being transmitted correctly. (Richard Modafferi points out that he has seen frequency-synthesized tuners with 10 kHz tuning increments. That is the increment needed to solve in every case the problem of an off-center station.) Off-center stations often occur on FM cable, when stations are moved around the dial to a different place than the over-the-air broadcast position. Provisions for connecting an oscilloscope to the tuner to check for multipath are also included. If you want to get depressed, try using this to view the quality of your cable FM on a scope. The antenna inputs are some nonstandard things that will take push-on connectors but not the screw-on type. This is a royal pain in the *tochis.*

The tuner automatically selects the stronger signal of the two antenna inputs. Signal strength is displayed in dBf. The measurements do not appear to be to extraordinarily accurate; although the signal-strength circuit is quite sophisticated, as explained below, our sample expanded changes for high-level signals and compressed changes for low-level signals. Even so, this is a very good way to get a feel for the relative strength of the signals. The tuner will automatically decide the control parameter, such as IF bandwidth, RF attenuation, blend activation, etc. It appears to do a reasonably good job, but you can override the tuner's choices and store your own choices if you wish. One nice feature that is not present on this tuner would be a recording calibration tone indicating the 50% modulation level. The record companies have apparently put enough of a scare into the hardware companies for the latter to drop any feature that would make things easy for the home tapist.

The RF section is the equivalent of six gangs. One is at the RF input and two are in the mixer section. The mixer is fully balanced. Another pair of tuned stages is in the local oscillator; one is used to set the oscillator's frequency, the other filters the oscillator as part of a buffer stage. The RF section can be bypassed through a separate tuning element that is coupled directly to the mixer stage. Local oscillator reradiation would appear to be a problem with this arrangement, but FCC regulations test for such things, and this unit could not be offered for sale if it failed the test. The ability to bypass the RF stage when hot signals are present is nice because RF overload problems due to inadequate dynamic range cannot occur if the stage is not in the circuit.

The IF stage for the wide and narrow modes starts with a single discrete stage and uses an integrated NEC µPC1163H amplifier between the ceramic filters. The final IF amplifier and limiter stage is the Sanyo LA1235, but the FM quadrature detector stage of the chip is used only for muting circuits. Two filters are used in the wide mode and four filters in the narrow mode. When the narrow mode is switched in, another µPCl 163H IF amplifier is put in the signal path. The supernarrow mode switches in a completely separate IF strip. One double-tuned LC stage, five ceramic filters, two discrete amplifiers, four µPC1163H integrated amplifiers, plus another LA1235 form the supernarrow IF strip. A diode switch selects which IF strip's output will be sent to the FM detector. Another µPCl 163H is used after this switch.

The signal-strength circuitry is quite complex. The Sanyo LA1235 has a signal-strength indicator output that is derived by measuring when its internal IF sections limit (it has six stages of amplification—see Figure 7). The earlier the stages limit, the stronger the signal. Unfortunately, even a relatively weak signal may limit the first stage because it has been through a significant amount of amplification in the IF strip that precedes the LA 1235. For that reason, signal-strength meters using the LA 1235 meter output are of limited usefulness. In the Onkyo, additional circuitry is included to test if earlier stages in the IF strip have gone into limiting. Additional places where limiting is tested for are at the output of the first amplifier in the IF strip, at an intermediate point in the supernarrow IF strip, and at the end of the supernarrow IF strip. The meter circuit has four—count 'em, 4—trim pots. Other tuners give you one. These are set for the correct indication at 5 dBf, 45 dBf, 85 dBf, and 105 dBf. Levels in between are less accurately displayed. A total of 50 active or passive components are used in signal-strength meter circuit. The muting circuit is also quite complex to prevent false mutes. This circuit monitors outputs from the LA 1235 chips in both the standard and supernarrow IF strips, as well from the PLL detector. A total of four adjustment pots are used in the circuit to insure it works reliably.

The FM detector is a PLL. This is the most nearly optimum FM demodulator, and that's what Onkyo needs to build a super tuner. The phase detector is the standard double balanced mixer topology made from a four-diode bridge and a pair of broadband transformers. Dual varactors and two MOSFETs form the VCO. The loop filter uses an NJM4560 op-amp. The other section of the op-amp also buffers the loop filter output and does the deemphasis equalization. This composite audio signal goes directly to the output stage in mono! It never sees the multiplex decoder. It is a sign that this tuner is intended for DXing deep fringe stuff, where the chance of getting anything usable in stereo is very unlikely. When stereo decoding is used, the loop filter output goes through a separate buffer circuit that includes the antibirdie filter.

Stereo decoding does not get as much attention as the rest of the design. An NEC µPC1223C multiplex decoder is used. The chip requires an antibirdie filter, as we saw above, and uses a low-end VCO design, namely an RC oscillator at 76 kHz that must be adjusted. An open-loop pilot-tone canceler is included, but no adjustments are provided to insure optimal cancellation. The age of the design really shows here, as much better multiplex decoder chips are available now. Individual channel-separation adjustments are provided for all three IF modes. Passive lowpass filters follow the decoder chip. The output buffer is an NJM4560. Two electrolytic capacitors are in the composite audio signal path in mono and four are used in stereo.

Construction quality of the unit is at the level of Japanese mass-market equipment; given all the stuff in the unit and the reasonable price, you would not expect anything else. The power supply to the audio section is ±15 V. A total of 8 voltage regulators is used. Adjustment of this tuner should be doable by any competent service technician because no distortion measurements are needed. The PLL detector does not need such adjustments.

The first unit sent to us by Onkyo performed very badly because it was either defective or totally misaligned. We sent it back. A second unit appeared to be working correctly and yielded the following test results: THD at 1 kHz in stereo was -62 dB in wide mode. That is very good, but it misses Onkyo's spec by 12 dB. The THD at 1 kHz was -36 dB in supernarrow mode. The Modafferi 10 kHz stereo IM test came out at -60 dB in wide mode, -64 dB in narrow mode, and -41 dB in supernarrow mode. Channel separation was >37 dB across the band in the wide mode. At 1 kHz it was 50 dB. That is 5 dB short of the specification given by Onkyo; the more important across-the-band separation figure is, on the other hand, 4 dB better than the specification. With the blend function enabled, channel separation is reduced to 18dB at 15 kHz. Frequency response in stereo fits into a ±0.4 dB window, which is better than the spec.

If you don't want to spend more than $1000 on a tuner and have difficult signal conditions, the Onkyo T-9090II is more likely to get you a listenable signal than any other tuner we have tested.

Rotel RHT-10

Rotel of America, P.O. Box 8, North Reading, MA 01864-0008. Voice; (800) 370-3741. Fax: (508) 664-4109. RHT-10 FM stereo tuner with remote control, $1499.90. Tested sample on loan from manufacturer.

Rotel and Harman Kardon are the only companies I know of that are using audiophile circuit-design attributes in the design of tuners. The Rotel RHT-10 takes things further than Harman Kardon has attempted. As we shall see, Rotel dropped the ball a couple of times, but I know of no other tuner that attempts to do what Rotel has done here. The RHT-10 is not designed to be a super tuner. Instead, the design is for signal conditions that are at least good.

In contrast to a spaceship like the Onkyo tuner, the Rotel has very few control buttons on the panel. Unfortunately, the presets are available only on the remote. Because of this I found this tuner to be a real pain in the neck to use (where did I put the remote??!!). Audio output is very high. It is fixed and must have been set here so the tuner would work with the Rotel RHC-10 passive control unit. It is a pain because the thing is 6 dB higher than CD standard level. You switch to tuner and then dive for the volume control on the preamp.

The RF section is the equivalent of six gangs, one at the RF input and two in the mixer section. The mixer is, surprisingly, not fully balanced. Another pair of tuned stages is in the local oscillator. One is used to set the oscillator's frequency, the other filters the oscillator as part of a buffer stage. As in the Onkyo, the RF section can be bypassed through a separate tuning element. How local oscillator reradiation is suppressed is unclear, but the FCC would not allow the unit to be sold if this problem existed. The IF strip starts with a single discrete stage and uses two integrated amplifiers between the ceramic filters. Two filters are used in the wide mode and four filters in the narrow mode. No super narrow mode is included. As I said above, this is not designed to be a super tuner. No instructions on how to adjust the tank circuit at the output of the mixer are included in the service manual. This adjustment typically involves a distortion measurement. Adjustment of components in the RF and IF stages can be done by any competent technician.

The last stage of the IF strip and the FM decoder is the el cheapo Sanyo LA 1235. For $1500 I thought I was going to get something other than a quadrature detector. Two adjustments are required for the detector, one involving a distortion measurement. The presence of the quadrature detector in this design is like a large zit on a pretty face because everything else is done so well for the reception of good-to-excellent incoming signals. The output of the detector passes through an antibirdie filter, and then the high-end fun begins.

At first blush things look normal, with a Sanyo LA3433 multiplex decoder chip on the board. It uses a 456 kHz ceramic filter in the VCO for narrow PLL lock range and low phase noise. But this chip is used only to generate the 19 kHz pilot tone for the pilot-tone canceler and the 38 kHz square-wave signal for the multiplex decoder. The rest of the circuits on the chip are not used at all. Instead, high-quality op-amps and discrete circuits are used wherever the audio signals actually travel.

First the 38 kHz output of the LA3433 is converted to a sine wave by means of a tuned circuit. This circuit is outside the PLL feedback loop, so adjustment of the tank is critical if the phase relationship between the composite FM signal and the 38 kHz sine wave are to be maintained. Two inductor adjustments do this. Pioneer puts the tuned circuit in the PLL loop when they generate a 38 kHz sine wave for their proprietary decoder. That eliminates the need to trim, but a custom IC design is required to do this. Rotel decided to do it with standard parts to save the cost of designing a special IC. Luckily, a simple level-peaking adjustment sets the inductors accurately, but they can drift after adjustment.

An NE5532AN chip (which is a lot better than what you will usually find in tuners' audio sections) is the active circuitry used to filter the 38 kHz sine wave. Another NE5532AN sums the incoming composite signal and the 19 kHz pilot tone together for pilot-tone cancellation. An inductor forming part of a bandpass filter that filters the pilot-tone canceler signal is adjustable The inductor sets the phase of the pilot tone, and a pot sets its amplitude. These are simple adjustments that can be made with a voltmeter, but again they can drift.

So now we have a composite signal with a pilot tone canceled and a 38 kHz sine wave. Multiply them together using a pair of analog multipliers and you would have birdie-free stereo. Unfortunately, low-distortion analog multiplier chips are expensive, so nobody has used this approach until now. Rotel does not use the analog multiplier chips, but they do take the plunge by making a discrete version of an analog multiplier instead. First the composite signal is converted into a pair of balanced currents. This is done with a clever two-transistor discrete circuit that uses feedback. These currents (I_{ee}) each enter the tails of two differential pairs. The 38 kHz sine wave signal (V_{id}) is applied to the bases of each differential pair. Now, the differential output current of a bipolar differential pair can be written as

$$I_{od} = k_1 I_{ee} \tanh(k_2 V_{id})$$

where k_1 and k_2 are constants. (I'll buy Bob Harley dinner at CES if he can tell me what k_1 and k_2 are.) For small values of V_{id} the hyperbolic tangent operation can be dropped (remember from senior math in high school something called the Taylor series), and now we have

$$I_{od} = k_1 k_2 I_{ee} V_{id}$$

Just what we want—an analog multiplier. Rotel uses an NE5532AN as a differential-to-single-ended converter at the multiplier core's output. The 5532 also acts as the current-to-voltage converter for the analog multiplier. Rotel uses the fully balanced structure to take care of even-order distortion terms in the multiplier core, like the errors caused by the presence of the hyperbolic tangent function. (OK, I hear you. This is a lot harder to understand than "the midrange is woolly," but you've got to admit it is more interesting *[...get a life, Dave!—Ed.]*, and it could actually affect the sound quality!) Channel-separation adjustment pots make sure the correct amount of L+R gets added to the correct amount of L-R to reproduce the left channel (or subtracted, for the right channel). Separate pots are supplied for the wide and narrow modes. The modes are switched by relays, not cheap transistors! The blend circuit that follows uses a CMOS switch, but the output muting is also done with a relay. It all sounds so very good, but something must be wrong under the surface because no antibirdie filter should be required in this setup, but it is in the signal path.

To conclude the signal path, audiophile-design-style AD847 op-amps are the output buffers in the Rotel. Electrolytics are used for the dc blocks at the front and rear of the buffer. The antibirdie filter circuitry adds three more electrolytics into the composite FM signal path. A second-order lowpass filter with a 15 kHz passband is formed around the AD847. Normally a higher-order filter would be used, and more high-frequency energy was observed at the output of this tuner than usual.

High-end design practice continues in the power supply. Separate secondaries of the large transformer go to separate bridges for the analog and digital sections. The analog side uses 4700 µF capacitors on the unregulated rails. A two-transistor open-loop regulator is driven by a reference formed with a current source and a zener diode to generate the regulated ±14 V rails for the stereo decoder. The LA 1235 gets a separate two-transistor discrete regulator that uses feedback. The other five regulators for the tuner are integrated units.

A very nice feature of this tuner is the inclusion of a separate IF strip (two ceramic filters and a discrete amplifier) plus an LA 1235 chip for the tuning meter. This is done because the main IF filter strip is designed to limit even on weak signals. Once this happens, the strength of the signal cannot be determined. That's why most signal-strength meters pin even when the tuner is receiving a relatively week signal. With this specially designed low-gain IF strip made just for the tuning meter, the relative signal strength of strong signals can be determined.

Construction and parts quality of the Rotel RHT-10, both inside and out, are a step above the Japanese mass market, but they are not quite up to the level of the Mcintosh MR7084 at the same price. For example, although a double-sided board is used, the top side is only a ground plane. The board has no through holes, and jumpers need to be used instead. Despite the build quality, some of the internal adjustments were way off. Channel separation was running 20 dB as the unit was delivered. The pots were noisy and had to be cleaned before they could be adjusted.

Although it is not designed as a super tuner—recall it has no supernarrow mode in the IF and that el cheapo quadrature detector causes the AM suppression to be no better than 60 dB—it sure performed like one in many ways. Real-world $2f_1 \pm f_2$ spurious signals were present only under the most challenging test conditions that Richard Modafferi could throw at it. Only the Accuphase beat it here, and performance was similar to that of the Onkyo T-9090II.

Selectivity was not as good as on tuners with a supernarrow IF strip but it comes very close. The Rotel can bring in things in narrow mode that require suppernarrow mode on the Onkyo. That means the signal will be cleaner on the Rotel, since the narrow filter distorts the phase less. But it must be noted that under worst-case conditions (40 dB differences in signal level between adjacent channels) the Onkyo could cleanly reproduce signals that were barely listenable on the Rotel.

THD at 1 kHz in stereo was -60 dB in wide mode and -44 dB in narrow mode. Better results in wide mode would have been possible with something a little better than the quadrature detector because the stereo decoder is designed to have very low distortion, as explained above. The Modafferi 10 kHz stereo IM test result was a superb -75 dB in the wide mode (no surprise, given the sophistication of the MPX stage). The IM result was -44 dB in the narrow mode. Channel separation after readjustment was 41 dB across the band in wide mode and 30 dB in narrow mode. Frequency response just made it inside the ±0.5 dB window given in the manufacturer's specification sheet.

At $1500 this is a good tuner. But at $750 it would be a runaway bargain for those of you who do not have worst-case signal conditions. But it is not $750—or is it? Rotel has a tuner called the RT-990BX that sells for just that price and is said to be almost identical to the RHT-10. We would like to tell you for sure, but Rotel of America appears to be no longer willing to lend equipment or even send a service manual to ***The Audio Critic.*** It would seem that any criticism of the company's products, no matter how slight and how well documented, is unacceptable to them. This once again proves that great engineering and production people can work for companies that have marketing employees with small-minded agendas and petty resentments. So if you have the cash and decent signal conditions, try the Rotel RHT-10. If you do not have money coming out of your ears but want to take a chance, try the RT-990BX.

* * *

The author wishes to thank Angelo Mastrocola and Richard Modafferi for reviewing the manuscript of the above article.

AES Technical Paper by Rich and Aczel

*On October 6, 1995, at the 99th Convention of the Audio Engineering Society in New York, David Rich presented to the Analog Electronics session his paper, coauthored by Peter Aczel, "**Topological Analysis of Consumer Audio Electronics: Another Approach to Show that Modern Audio Electronics Are Acoustically Transparent.**" The preprint number of the paper is **4053.** Copies are obtainable from the Audio Engineering Society, 60 East 42nd Street, Room 2520, New York, NY 10165-2520. Voice: (212) 661-8528. Fax: (212) 682-0477.*

Indoor Antennas and Boosters for FM

By David A. Rich, Ph.D.
Contributing Technical Editor

A little bit of this material may overlap the main article on FM tuners but is repeated here in context for the sake of clarity.

Old-timers who are into FM may recall Leonard Feldman's test of 11 outdoor antennas in the January 1983 issue of *Audio*. The tests were run at the CBS Technology Center. A testing tower 75 feet above ground level and 400 feet away from the source antenna was erected. The relevant specifications of the antennas were then measured. Needless to say, we do not have such a tower set up here at ***The Audio Critic***, nor do we have the RF test equipment to conduct the tests even if we did have the tower. So this report on indoor antennas is going to be mostly a use test.

OK, let's start with the traditional disclaimer that if you want the best performance from your FM receiver you want a roof antenna. First, it is obvious that on the roof you have the advantage of additional signal strength through height. In addition, you have the advantage of preventing the antenna from interacting with building materials, stuff in the room, or people. Needless to say, aluminum siding makes a great Faraday shield. But I hear you whine that installing an outdoor antenna is expensive. Just think of it this way: a $300 antenna installation driving a $300 tuner is going to work a lot better than a $600 tuner connected to a cheap indoor antenna. The reason is that the fundamental limitations of an FM receiver require that the signal at the antenna input be free of noise and multipath if the tuner is to operate at maximum performance. So why are reviewing all these indoor antennas? In a single word, apartments.

Let's start by discussing what we should have measured if we had that tall tower. The first important test is the antenna gain. Gain is measured relative to an omnidirectional antenna of proper size for the test signal's transmission frequency. Gain is measured at the main lobe of the polar pattern of the antenna's reception. To reject interfering adjacent or cochannel signals that originate from other specially separated transmission towers, or to reject echo signals (multipath), we want the antenna to attenuate signals not at the main lobe. Two specifications summarize the characteristics of the reception polar pattern. One is the beam width, which is the angle around the main lobe for which the gain of the antenna stays above -3 dB (since we are working with signal power, -3 dB is the half-power point) relative to the gain at the main lobe. We want the antenna to be as directional as possible, so we want the beam width to be as small as possible to reject interference. Of course, if the beam width is very narrow, it might become impossible to set the antenna's main lobe in the direction of the station even with a rotor. The other specification that describes the polar pattern is the front-to-back gain. The name says it all, since it's the ratio of the antenna's gain at its main lobe to the gain 180 degrees from the main lobe. This again represents a figure of merit as to how well an undesired signal will be rejected. An interesting phenomenon that results when an antenna has a nonuniform reception pattern is passive gain. Passive gain is best explained by an example supplied by AudioPrism's Victor Tiscareno. Imagine a light bulb in the middle of the room. All areas of the room will receive light of equal intensity. Now put a reflector in the back of the bulb. Now the light from the front of the bulb is increased, but at the expense of the light in the back of the bulb. Similarly, an antenna with a narrow beam width will have gain enhancement at the center of the main lobe relative to an omnidirectional antenna.

The next important specification is the antenna's output impedance. For maximum power transfer, we want the antenna to present a purely restive load equal to the input impedance of the tuner (75 or 300 ohms). Uh-oh, now I did it. The audio cable manufacturers have all risen to their feet to tell us about the characteristic impedance of their cable. Please sit down, folks; this maximum power transfer stuff is for RF systems, and I do not care at all about the transmission-line characteristics of your audio cable.

Real antennas do not present a purely restive load at all frequencies, and the impedance that the antennas do present varies with frequency. When the antenna does not present the ideal termination impedance, some of the energy received by the antenna is reradiated out of the antenna (the audio cable folks are rumbling again—forget it, the wavelengths of the signals in the audio band are so much larger than at RF that this effect is irrelevant in the audio band). The measure of this phenomenon is called the voltage standing-wave ratio (VSWR) because it is the ratio of the maximum voltage of a signal in a transmission line to the minimum signal level in the line. If the line is perfectly terminated , the VSWR is unity.

The worse the termination, the higher the VSWR. The worst-case value of the VSWR over the band of interest (88 MHz to 108 MHz) thus presents in a single number the accuracy of the impedance match of a given antenna.

So what is wrong with the two-dollar dipole that came with your tuner? It is actually not as bad as it is made out to be if it is aimed properly. One problem is the that the dipole is bidirectional, so it does not reject a multipath signal coming from the rear. (The nonuniform polar pattern does give the antenna 3 dB of gain.) But the really big problem with the dipole is that, unless your wall just happens to face in the exact direction to maximize the signal pickup of the station you want, the dipole will not do a good job. Even if you luck out with one station, if you want multiple stations you are almost certainly out of luck. That is were the **Magnum Dynalab Silver Ribbon** comes in.

Essentially the Silver Ribbon is a rabbit-ear antenna. The difference is that the single rod on each side of the standard rabbit-ear antenna is replaced by a U-shaped element formed by parallel metal bands connected at the end. The size of the bands can be adjusted. The instruction manual claims that maximum reception results for the FM broadcast band with the elements fully extended, but reducing their size will sometimes improve the impedance match of the antenna. The adjustment of the antenna is made while watching the signal strength meter. The elements are adjusted for maximum signal strength. Unlike a dipole, this antenna can be easily rotated for optimal signal reception. In an area of high or moderate signal strength, it may be all the antenna you need. My experience with it showed it performed well in bringing in strong signals when properly aimed. Like most rabbit-ear designs, it is somewhat sensitive to the presence of the person fiddling with it. I was unable to figure out how to rotate it without being near it. On weaker stations a properly aimed dipole could outperform the Silver Ribbon, probably because of the height advantage gained by mounting the dipole close to the ceiling. In addition, the rabbit-ear type of antenna is usually near metal objects, which can affect its performance. Unfortunately, a properly aimed dipole is a mater of luck.

Moving up a step in price, and with significantly improved cosmetics, we have the **AudioPrism 6500.** Inside the small 9.5 by 9.5 by 2.5-inch box is a loop antenna. The use of the loop antenna allows the size reduction from the standard dipole, but this type of antenna has a high Q. As Richard Modafferi points out, it is basically a big LC tank circuit in a box. This means that the antenna is sensitive to only a small range of signal frequencies, and thus it must be tuned. A small knob on the front of the box is used for the tuning. That is good news if you have a selectivity problem with your tuner. Such a problem is most likely to occur when the station you are trying to tune in is weak and a strong, relatively close signal is present. The 6500 even allows you to try and capture a weak signal by placing a cheap Radio Shack amplifier after the antenna. Without a high Q antenna, the Radio Shack unit is usually overdriven by the strongest signal on the dial and makes things worse, not better. The same problem occurs with the teeny active antennas that are often sold by dealers instead of the better passive antennas, such as I am reviewing here. (Have you ever seen a good review of one of these weensy boosted antennas? Apparently they sell because of the cute plastic case or the fact they are plugged in—the unwashed masses think if it has active electronics it must be better—or the very high markup on this class of products.) Do not think you have built an el cheapo substitute for the Magnum Dynalab "Signal Sleuth" (reviewed in this issue) by combining the 6500 and a Radio Shack amplifier this way. The bandwidth of the 6500 is pretty broad, since the -3 dB points are 1.2 MHz apart and the transition band is shallow because this is a single-tuned system.

The downside of the 6500 is that if you do not need the selectivity you are stuck fiddling with this antenna every time you change a station. The VSWR of the antenna when tuned is clamed to be 2.1:1 or less.

An interesting feature of this antenna is that it is omnidirectional when placed flat on a table or shelf, but when it is set upright on its narrow side it becomes directional. When it is stood up on its side it is clamed that the radiation polar pattern is cardioid in shape. Front to back gain is clamed to be 6 dB. Since it is now directional, it picks up about 3 dB of gain. If you are in an area with strong, interference-free signals, you can leave the antenna flat and do not have to worry about moving it around when changing stations (you still have to tune it—drat). If you have multipath or a strong undesired signal, the 6500 may yield better reception than a dipole or rabbit ears because it does have some front-to-back rejection. Things get a little more unpredictable, though, because the antenna receives the horizontally polarized component of the signal when it is flat but the vertical signal component when it is on its side.

When the signal strength was high enough, the 6500 worked very well in my setup. It almost always equaled or outperformed a dipole or rabbit ears. Best results occurred most often with the antenna on its side. Again, a properly aimed dipole near the ceiling performed better on weaker signals, for the reasons cited above. Adding the Radio Shack amplifier or even the Signal Sleuth did not help. AudioPrism never claimed the 6500 was designed for a fringe-area signal; that is why they make other antennas.

If you want to bring in weak stations, you need something bigger with more gain, like the **AudioPrism 7500.** Did I say bigger? Well, this thing is seven and a half feet tall! That is how big the enclosure has to be to house a full-size half-wavelength monopole antenna on a quarter-wavelength matching stub. The antenna is covered with a four-inch diameter tube. Its base is a 13-inch diameter wooden disk, which also contains a foil ground plane. The ground plane attempts to isolate the coaxial cable exiting from the antenna, so it will not interact with the antenna's radiation pattern. For this to be completely effective, the ground plane would have to be a half wavelength long, but that is clearly impractical.

Metal spikes attached to the base keep the unit stable. The unit has a low WAF (Wife Acceptance Factor—perhaps the PC crowd would want to change that to SAF for Spouse Acceptance Factor), which could get even lower if the spiked feet scratch an expensive hardwood floor. AudioPrism will apply custom fabric to the 7500 if this is required to bring about spouse acceptance. Please note that the 7500 also has the potential to increase the MAF (Mortgage Accumulation Factor), since your spouse can bring up the removal of the 7500 and its replacement by a roof antenna as one more reason to leave the apartment and purchase a house.

Now, the first thing that you ask about this antenna is how does it work, since it is a vertical pole, not the horizontal rod that we associate with FM and TV antennas. The answer comes from the fact that electromagnetic waves can be polarized, just like light waves. Originally, FM stations broadcasted only a horizontally polarized signal, hence the need for a horizontally oriented antenna. With the advent of car FM radios, things had to change, since a car antenna is a vertical device. Radio stations started to broadcast circularly polarized signals in response to this need. The 7500 takes advantage of this, and it is thus designed to receive the vertically polarized component of the transmitted signal.

The 7500 is claimed to have a suppressed vertical radiation pattern. This is achieved by adding a quarter-wave matching stub to the half-wave element. This apparently is called a J-pole in amateur radio circles. (Unlike modern audiophiles, the amateur radio crowd has had real problems to deal with. The origins of this design are said to date back to developments during World War II, according to Victor Tiscareno.) The radiation patterns of each of the elements differ, and using both elements in this configuration sets the radiation pattern of the antenna to see more energy coming from the horizon. This modification of the radiation pattern also allows the antenna to have a gain of 5 dB. The radiation pattern also helps reduce the antenna's sensitivity to multipath reflections in the vertical plane, such as those created by the overflights of airplanes. Of course, since this an omnidirectional antenna, it is much more sensitive to multipath than a directional unit. It is also unable to reject strong adjacent- or alternate-channel stations that are spatially separated from the desired station. The VSWR of the antenna is clamed to have a worst-case figure of 1.9:1. This figure, which is as good as that of almost any outdoor antenna, is still not as good as claimed for the **Day Sequerra FM Urban Antenna,** which is very similar to but twice the price of the 7500. The VSWR of the Urban Antenna is claimed to be 1.4:1. Apparently the 1.9:1 is a guard-banded figure, and Mr. Tiscareno informs us the antenna typically has a VSWR of 1.5:1. We did not get to test the Urban Antenna, but we should note that it must be wall-mounted, whereas the 7500 is free-standing, and the Urban Antenna does not have the ground plane.

I expected good things from the 7500 but I did not get them. I tried it out in a number of test locations, even though lugging this seven-and-a-half-foot thing around is not a lot of fun. (You keep bumping into things, or it gets tipped over. Needless to say, the antenna had a few bruise marks on it when we returned it.) In every test location the 7500 was outperformed by the 6500 or the APPA-8500 (see below), and even the lowly dipole. My guess is that the antenna's omnidirectional characteristic was the problem. I can confirm this thing has gain, since it could produce some wild DXs (I got stations broadcasting over 100 miles away—most likely atmospheric scatter), especially when the Signal Sleuth was in the signal path. Unfortunately, there was a lot of fading on the weak signals; I would not say any of these distant stations was clean enough for serious music listening. Please understand that none of the above should reflect badly on this antenna. Others in different areas of the country have reported very satisfactory results with the 7500. (For example, see Fred Rosenberg's column in "Issue Number Five" of the now defunct *Sounds Like....* I must add that Rosenberg's columns in *Sounds Like...* were the only thing there worth reading.) Victor Tiscareno reports that the 7500 works best in flat areas that do not have multipath problems. Kansas residents, please take note. But what I needed was gain and directionality. What I needed was an **AudioPrism APPA-8500.**

AudioPrism calls the APPA-8500 "the ultimate FM antenna system...the first multi-element, directional and full-sized indoor FM antenna on the market today..." It's hard to argue with this; the antenna has three selectable lobes with 120° beam width, a front-to-back ratio of 10 dB, a gain of 8 dB, and a VSWR of 1.9:1. A good outdoor antenna would have better specs, with less than half the beam width and a front-to-back ratio of better than 16 dB, but the APPA-8500 comes closer to replicating the performance of an outdoor antenna than anything else known to me. The problem is that it is inside the building and not up on the roof, and that makes all the difference in the world.

The APPA-8500 is not easy to hide: it is 63 inches tall and 12 inches in diameter. Inside are three half-wavelength dipole antennas. Directionality is achieved by using two of these antennas as reflectors for the third active one. In my opinion, the 8500 has a higher WAF than the 7500 (its proportions make it look a lot less weird), but it retains the same spiked feet that could increase the DPF (Divorce Potential Factor) or LBF (Lease Breakage Factor—remember, if you live in a house you want an antenna on the roof) if they end up doing damage to a wood floor. Optional oak end caps are offered to increase the WAF.

A wired remote control with a 20-foot cable allows you to select one of the three antenna lobes, or you can select all the antennas in an omnidirectional mode. The box does this by determining which antennas act as reflectors and which is the active element. In the omni mode all three are active. The big advantage of this remote approach is that you are not near the antenna; thus you are not affecting it by your proximity to it. The remote box also allows for selectable attenuation of the signal by as much as 18 dB. The box is line powered, but this is only to drive the relays which select the antennas. No electronic amplification is used.

All this high tech did great things for me. The antenna's gain brought in the weak signals, and the directional design of the unit kept out interference sources. The omni position always resulted in increased background noise. This showed why the 7500 did not work in my location. I wrote out a check to the manufacturer to purchase the 8500 the second day it was in my apartment. After that it never left the apartment. I was afraid that if the Editor got hold of it he would never give it back. *[You forget, Dave: I live in a house.—Ed.]* That said, this does not mean you should rush out and purchase one. If signal strength is not a problem, the less expensive directional antennas may give as good results as this five-foot 25-pound monster.

The key thing I want you to come away with is that all these indoor antennas are intelligent designs and good values, but because of their diverse design goals the only way to find out which will work best for you is to try one. I hope the information above will help you narrow down the choices. Never purchase an antenna without a money-back guarantee. Your dealer should be able to help you choose an antenna, but most dealers want to have nothing to do with antennas because of the strong possibility that a given antenna will be returned. Selling overpriced audio cable is much easier and more profitable.

Finally, I restate that none of these antennas will ever come close to matching the performance of a good outdoor antenna. If you can install such an antenna, by all means do so. It will be the cheapest way to upgrade the quality of your FM signal.

AudioPrism, 1420 NW Gilman Boulevard, Suite 2593, Issaquah, WA 98027. Voice: (206) 641-7439. Fax: (206) 644-5485. High-Q 6500 indoor antenna, $125 (wood), $99 (vinyl). Model 7500 indoor antenna, $299. APPA-8500 indoor antenna, $499 (w. remote).

All three AudioPrism antenna models are reviewed in the article above.

Magnum Dynalab 205 'Signal Sleuth'

Magnum Dynalab Ltd., 8 Strathearn Avenue, Unit 9, Brampton, Ont., Canada L6T4L9. Voice: (905) 791-5888. Fax: (905) 791-5583. Magnum Dynalab Corp., 1237 East Main Street, Bldg. #2, Rochester, NY 14609. Voice: (716) 654-6340. Fax: (716) 482-8859. 'Signal Sleuth' 205 FM antenna signal amplifier, $279. ('Silver Ribbon' indoor antenna reviewed with related products in article above.) Tested samples on loan from manufacturer.

The world of high-end audio is a strange place. Dozens of companies produce nothing but overpriced interconnect cables. The audiophile can choose among 1000 different versions of these useless products. On the other hand, if you need a tunable FM antenna amplifer, a genuinely useful product, you have a choice of one. Luckily it turns out to be a good one.

But why would you need one? Its primary use is to amplify low-level signals so that the signal level at the FM detector will be high enough to insure low-noise reception. So why can't you just buy a Radio Shack amplifier for $30.00? First, the Radio Shack unit is not tunable. Most weak stations are surrounded by stronger signals. With an untuned amplifer you raise the signal level of both the weak and the adjacent strong stations. If the strong station does not cause the amplifier to overload, it will cause the tuner to overload. The Signal Sleuth has a tunable 600 kHz bandpass filter. Thus even alternate channels (400 kHz away from the carrier frequency of the desired station) are reduced in amplitude by the Signal Sleuth. (This is not to be confused with an adjacent channel, which is 200 kHz away from the carrier frequency. Note that a station occupies a spectrum with a width of 200 kHz about its center frequency. Thus, to attenuate an adjacent channel, significant filtering must occur 100 kHz away from the center frequency of the desired station.) Second, the Radio Shack unit has significant self-noise and is relatively nonlinear, which can cause spurious response problems (see below). The result is that the signal coming out of the unit may be more corrupted than the signal coming in. Note that small amplified antennas have exactly the same problem.

It is important to understand that the filtering function of the Signal Sleuth does more than just reject some of the alternate channel. It also improves image rejection. In a superheterodyne receiver, the incoming RF signal is mixed with a signal from the local oscillator. By changing the local oscillator frequency for each station, the output of the mixer becomes a signal which has the same center frequency for all stations. A problem with this is that the mixer can translate signals both above and below the local oscillator frequency. It is thus possible for an undesired signal to get translated to the IF frequency. For most FM receivers the local oscillator is set above the incoming RF signal; thus the image frequency is the desired RF signal's frequency plus two times the IF frequency. The IF frequency of most FM receivers is 10.7 MHz. Note that the image signal is never a broadcast FM signal. Other image interference is possible because the mixing signal often contains harmonics. These harmonics also generate sum and difference products that can move an undesired signal to the IF frequency. Filtering the RF signal prevents these image signals from contaminating the desired signal by attenuating the undesired signal before it gets to mixer.

A second problem, resulting in other spurious responses, is the nonlinearity of the RF amplifer or mixer. This can cause intermodulation products, which can again fall into the IF band. Under worst-case conditions, with a heavily overloaded RF front end, a station may appear at a number of places on the dial. A common mechanism for a spurious signal is the second harmonic of the local oscillator beating against the second harmonic of an RF signal. The Signal Sleuth again helps by filtering out undesired signals before they get to the input of the tuner. In the case of a desired signal that is very strong, the Sleuth provides attenuation to prevent overload of the tuner. In the good old days when super tuners had super RF stages, you would not have needed the Signal Sleuth. For example, the Technics ST-9030 had an eight-gang tuning capacitor and two RF amplifier stages. It had an image and spurious-response rejection of 135 dB. Today you get one RF amplifer and the equivalent of three or five gangs. The result is that some of today's best tuners have image and spurious-response ratios of only 80 dB. Unfortunately, if you have two stations close together that cause a spurious signal, then the Signal Sleuth cannot help. Such a spurious response is caused by the RF stage or mixer stage in the front end. These circuits generate the intermodulation-distortion products that cause the spurious response. Adding more stages of RF as the Signal Sleuth does may actually make things worse.

The Signal Sleuth cabinet is the size of a small preamp. The unit looks right at home in an equipment cabinet with other high-end stuff, with its thick, black, anodized metal faceplate and aluminum knobs. Two switches are on the left of the front panel, one for power, the other for bypassing the unit. Of the two knobs on the right of the front panel, one controls the gain or loss of the unit, the other tunes the center frequency of the bandpass filter. The gain control sets the gain from -30 dB to +30 dB. It has a click position when rotated fully clockwise; this is used to tune the unit. With the gain control in the click position the tuning knob is adjusted for maximum signal level as read on the tuner. Then the gain control is adjusted to a point where the received signal quality does not improve with increasing rotation. Putting more gain in the path than is necessary can cause overload problems in the tuner's front end, especially if strong signals are near the weak signal you are trying to receive.

Inside the unit is a single-sided PC board, a small transformer, and the front-panel-mounted controls. On the PC board is a three-stage RF amplifer. A tuned LC circuit is used in each RF stage. The stage is tuned using a varactor diode in place of the capacitor. Varying the reverse bias voltage of a varactor diode causes its capacitance to change in a predicable fashion. The front-panel control sets the voltage on the diode, and this is how the unit is tuned. One would think that Magnum Dynalab would provide an external dc input so that their tuners could send the dc control voltage to the Signal Sleuth. That way Magnum Dynalab tuners could control the Signal Sleuth and it would not have to be tuned separately. Such an option does not exist on this model. The front-panel switches and tuning knob are sealed units of good quality. The gain pot is a much lower-quality unit. The gain adjustment as well as the bypass function are also accomplished by dc control. The RF signals never leave the PC board.

Some tuners have circuitry which selects IF bandwidth and stereo blending functions automatically, based on signal strength. Often a clean but weak signal is degraded by this process. An interesting thing you can do with the Signal Sleuth is to gain a signal up to the point where the tuner goes into wide-bandwidth mode and turns the blend off. Of course, it does not make sense to put a $250 RF amplifier in front of a $200 tuner so you can defeat some of its automatic switching circuits. A better approach would be to sell the tuner and purchase one with manual override.

An important thing to understand about an RF amplifer is that it is really effective only if a weak signal is not embedded in noise. If the signal is in the deep fringe zone, or if you are using a small indoor antenna, this may not be the case. If the signal is embedded in noise, all you are going to do is to gain up the signal *and* the noise. This will accomplish nothing. To see if the Signal Sleuth will help you, it is necessary to try it. Your dealer should be willing to loan you a unit. (See why they would rather sell wire!) The Signal Sleuth probably performs at its best when connected to a good high-gain roof antenna with a rotor. This setup would give the highest signal-to-noise ratio at the input of the Signal Sleuth. Using a small indoor antenna is probably not going to work so well. I tested the Signal Sleuth with a setup midway between these two, using the AudioPrism 8500. In some cases the Signal Sleuth did nothing to improve the signal, but it was able to deliver one previously unlistenable NPR station in decent mono and it improved another NPR station from strictly mono to somewhat noisy stereo. It worked well enough for me to make me decide to buy it. •

Commodities and Revivals

(It Don't Need Revivin' Unless It's Already Dead)

By Tom Nousaine
(The High-Definition Weasel)

"LPs Making a Comeback" and "Tubes Breathe with New Life" are headlines we see with a fair degree of regularity these days. Is it true? Do these ancient technologies have forgotten performance advantages that all the world's top designers and scientists have managed to overlook in the race to digital and microelectronics?

Of course not. Why the hoopla then? Why are we reading about revivals in the consumer and underground press? The answer involves economics. Ever heard of the Product Life Cycle? It is a forecasting tool that predicts sales of a product (and the underlying technology) over a growth-and-decline cycle with several definable stages. Understanding the PLC will help us understand the market dynamics that enable revivals.

A new technology must potentially offer the means to produce goods an order of magnitude better than existing products to capture financing for startup. Existing products often have legions of loyal users, so a technology with zero market share must promise fundamentally superior performance to warrant consideration.

To complicate things, first generations of the new technology often need considerable debugging. For example, early transistors were pretty unreliable. Furthermore, the "old" technology often suddenly gets much better when a successor appears on the horizon. Remember how, after being "too expensive" for 15 years, linear-tracking tone arms miraculously showed up on even close-'n' -play turntables?

After the Introduction Phase, sales and revenues skyrocket as the newer technology continues developing and the market recognizes the innate performance/cost advantages. People stand in line—they stood in line to buy CDs and players in the mid-80s during the Rapid Growth phase of the life cycle. They are standing in line to buy DSS now.

Eventually, growth slows as the market Matures and Saturates. Products take on the characteristics of a commodity as performance becomes optimized. Promotion, style, features, and price become the major competitive forces in a Mature market.

Sooner or later, someone finds an even better way to provide the function and, as a new contender enters the Introduction and Rapid Growth stages, the now "old" technology slides into Market Decline, although sales stagger along for a while as prices plummet and inventories are "blown out."

So what happens after the Market Decline? Because the production facitities have long since been paid for and are generally still in pretty good working order, producers work like hell to squeeze a few more dollars out of them. Products fall into specialty market niches, where demand from a small number of people who have strong attachments, often sentimental, to the older technology remains strong.

In this Residual Use phase, marketing becomes paramount. Manufacturers find increasingly clever ways to "resell" output to a small, dwindling base of customers. "Revivals" and press announcements proclaimimg the "Second Coming" are major promotional tools used to help squeeze the remaining life out of an obsolete technology.

Prices often increase dramatically, even though fixed costs of production are low because the plants were fully amortized during the Growth and Maturation phases. Variable costs inevitably increase as smaller and smaller lots of raw materials are being bought and each promotional dollar sells fewer units. However, loyal followers display price-inelastic behavior—they continue to buy even when prices rise because their main attachment to the products is sentiment- and not performance-driven.

The producers' main job becomes maintaining contact with the old users and attractively promoting the product. Pitches aimed at "exclusivity" and "connoisseurship" are extremely effective at this stage.

Sometimes seemingly unrelated economic events help out producers. For example, following the Cold War, Soviet military cutbacks left many overseas vacuum-tube producers with excess capacity and a corresponding need to find new customers. Viva! Revival in North America!!

It is easy to forget that it was the superior performance of the new technology that sucked out demand for the old technology in the first place. Repeat: the new stuff had to be radically superior to the old to enable displacement of an entrenched market.

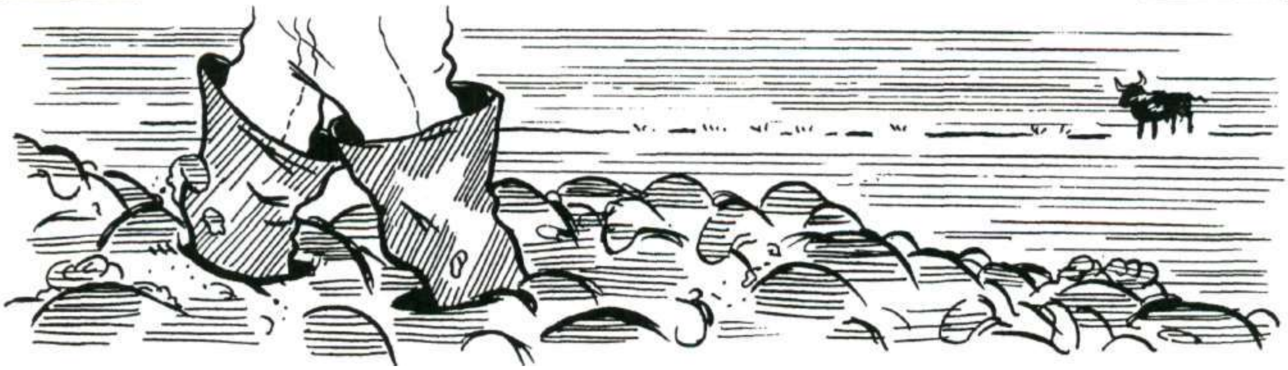
There are false starts too. Quad, for example—a new technology that was not well enough developed to displace the existing products. Furthermore, most major audio advancements are tied to developments in other markets. For example, the development of digital audio products coattailed advancements in telecommunications and computers. Of course, we were all dragged kicking and screaming into multichannel audio (including stereo) by the movie people.

Solid-state electronics and CDs have already reached the Market Saturation stage, where they have basically become commodities. Power amplifiers and CDs (excuse me while I duck) are so competent that they are basically differentiated by Price, Features, Style, and Promotion. Even the goldenest of ears cannot tell an el cheapo transistor piece of junk from a pair of the best monoblock power amplifiers ever built when the blindfolds come out. Same with preamps and CD players.

Other signs of Market Saturation are obvious when you know what to look for. Remember the Watt Races and the Distortion Races where the battles raged at specification levels that had zero performance impact? Today's Jitter Race and Bit Wars (Super Bit Mapping, HDCD, outboard DACs, Legato Link, etc.) are good examples of marketing strategy based on style masquerading as per-

Hip Boots

Wading through the Mire of Misinformation in the Audio Press



Editor's Note: This is the third issue without David Rich's byline in this column but not because he has lost interest in audio mendacities and techno-howlers. On the contrary, he remains the chief instigator here, constantly finding new outrages and calling me up to sic me on the offenders. It's just that the limited time he is able to spend on The Audio Critic *must be allotted to circuit evaluation and equipment reviewing. The steady broadening of the "Hip Boots" marshes and swamps now makes me think that an overview of their "geography" is in order, with capsule summaries of the principal sources of polluted information. Call it an abstract of our demonology.*

As I have repeatedly discussed and explained before, there exists an unfortunate schism in today's audio community. On one side, there are the scientific audio practitioners, consisting of (1) members of professional engineering societies such as the AES, the IEEE, the ASA, etc.; (2) high-level engineering employees of large electronics companies such as Sony, Philips, AT&T, etc., responsible for important product development; (3) academics in the E.E. and physics departments of universities; and (4) a few technologically disciplined audio journalists, whom you can probably count on your ten fingers. Of course, (2) through (4) often overlap with (1). On the other side, there are the antiscientific tweaks and cultists, who may or may not earn their living in audio but who never, never have graduate degrees in engineering or physics and who are never responsible for product development anywhere above the trendy boutique level. A lot of them write about audio. None of them give a hoot about scientific proof. All of them influence the innocent.

Let us zero in on the principal habitats of this antiscience faction.

The high-end dealers.

There are only a few hundred of these from coast to coast, but they set the tone. A well-intentioned but poorly informed music lover walks in with a bundle of money, and with just a little bit of bad luck he/she will end up with a 9-watt single-ended triode amplifier and silver cable. I'm not saying there isn't a high-end dealer in the country who will recommend sensible equipment and good value, but there can be no doubt that the audio salons are the main incubators of costly misinformation.

No, I'm not going blame the high-end manufactur-

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formance.

In all of these cases products are essentially differentiated with the equivalent of chrome bumpers and trim, while producers proclaim fantastic improvements in sound. Interestingly, some of the fiercest advocates are the very people who mocked the big Japanese participants during the Distortion Wars.

An audio-enthusiast performance fanatic (a.k.a. Weasel or Geak) deploys his resources where they return the biggest payback in sound quality. After cable, interconnect, and DAC differences evaporate when the blinders come out, a Geak looks to Better Recordings, New Formats, and Better (and more) Speakers to deliver more music.

Notice I didn't say "more music for the buck." Resources, time, as well as money, are finite. Even if you can afford it, wasting time auditioning cables and DACs suboptimizes the throughput of your system. There are more effective ways to improve the sound of your system.

What goes around, comes around. Be sure you keep an objective eye open when the salesman with the Revivalist Hat appears at your door. Not that buying old technology is necessarily a bad thing. On the contrary, you may need a new record player to retrieve certain program material that may never appear on CD. Just don't kid yourself into thinking that a new record deck or an outboard DAC is somehow going to magically make your system sound better.

Is there a Tube Revival and LP Revival? Sure, but only a dead technology needs Resurrection. 0

ers, at least not much. They will make whatever sells and stop making whatever doesn't sell. The high-end market is driven by the beliefs and dreams of the high-end customer, who in turn is putty in the hands of a know-it-all floor salesman. It's him that the manufacturer must impress and cultivate, not the buying public. The savviest high-end manufacturers have always known that.

Stereophile magazine.

This is the tweako journal currently preferred by the true believers. According to one highly intelligent and ethical manufacturer I happened to discuss it with, it is addressed mainly to the floor salesman, who always needs highfalutin technical arguments for one-upmanship and customer manipulation. I don't think there are enough floor salesmen in audio to account for the magazine's entire circulation, but the characterization is on target. It's certainly not the engineering/academic community that supports *Stereophile;* I hear only snickers from degreed professionals when the name comes up, if not vehement contempt.

What's truly insidious about this publication is that they use electronic measurements *cosmetically,* just to create a *visual* aura of scientific objectivity, without the slightest effort to link the readouts, graphs, charts, etc., to their totally irresponsible subjective conclusions. They are opposed to controlled (ABX) listening tests, which of course do not serve their agenda, and come up with the most outrageous pseudoscientific sophistries to reject the overwhelming evidence of such tests as performed by others. Tom Nousaine has a large collection of case histories on that subject.

Perhaps the most infuriating of their smoke-and-mirrors bench tests are Robert Harley's pretentious and wrongheaded jitter measurements. Bob Adams of Analog Devices, Inc., one of the world's most highly respected authorities on digital jitter, has repeatedly explained in simple language (in our Issue No. 21 and elsewhere) why you cannot measure jitter the way Harley does. Harley and the *Stereophile* editors know this, yet he keeps doing it the same old untutored way, and they keep publishing it. Even the Catholic doctrine of the forgiveness of sins excludes *obstinate* sinning; Harley's obstinate perversion of digital theory is surely unforgivable in the secular here and now.

Hi-Fi News & Record Review.

American culture has its oldest roots in England and so does the tweako culture of Santa Fe's *Stereophile. HFN/RR* is the leading antiscientific audio journal in the U.K. and the former editorial demesne of John Atkinson. That's where he developed the reviewing style and journalistic attitudes that originally attracted *Stereophile* publisher Larry Archibald's attention. He hasn't been with his alma mater for almost ten years but it seems to have made no difference; the intellectual climate there remains the same because unaccountability is a habit that tends to linger. Everything in *HFN/RR* must be read exactly as if it appeared in *Stereophile*—with several grains of salt.

The Absolute Sound.

Harry Pearson, Editor and Publisher of this ultra-tweako journal, is the Charles Manson of audio. Not that he instigates homicide, but consider all these similarities: he is totally self-absorbed; he spouts muddle-headed philosophy at the drop of a hat; he has delusions of grandeur (the "Pearson Publishing Empire," etc.); he has a small but fanatical pack of disciples who will commit violence (to science, logic, common sense, even human decency) at his bidding; when provoked he rants and raves and lashes out, completely out of control; faithful cohorts suddenly turn against him—I could go on but I think the point is made.

Yes, he was one of the mid wives of the High End more than twenty years ago, but he never really understood the cause-and-effect relationship between technology and the sounds he was hearing; he believed, and still does, that a good amplifier is made like a violin by some kind of sensitive Stradivari-like artist/craftsman. A good general-science teacher in 9th grade would have done him a world of good. Today his magazine is a farrago of tubes, vinyl, phono cartridges, and interconnects—so deeply immersed in weirdness and unaccountability that not even tweaks can take it seriously anymore. Every once in a while he feels the need to run a "technical" article; since no practitioner with genuine credentials wants to have anything to do with *TAS,* he has to get pseudo-techies like the Canadian charlatan Gerard Rejskind (see this column in Issue No. 19) to do their voodoo. New "brilliant" technical writers have been announced—yeah, right, and Ralph Nader is joining General Motors.

Audio magazine.

This is *the* original audio publication, the granddaddy of them all, founded as *Audio Engineering* in 1947. The "engineering" part, eventually dropped from the name, was completely dominant for many years, although the *Journal of the Audio Engineering Society,* launched in 1953, then became the official publication of the audio engineering profession. Sometime in the mid-'80s, long after Gene Pitts had become the editor, *Audio* changed its editorial course and began to give "equal time" to science and fantasy. Alongside such thoroughly solid engineer-reviewers as Don Keele and David Clark, there appeared unaccountable golden-eared dilletantes like Anthony Cordesman (blithely commuting from *TAS)* and Jekyll-and-Hyde techie/tweaky split personalities like Bascom King. Even the late Len Feldman, a rock solid laboratory practitioner for decades, began to hear things that didn't exist. Before you could believe a review in this strange new something-for-all-tastes *Audio,* you had to know your players.

The irony is that Gene Pitts, who had presided over this bastardization process from its inception—I don't know exactly how willingly or reluctantly—has recently been let go and replaced by Michael Riggs, former executive editor of *Stereo Review,* whose audio philosophy is thoroughly objectivistic, just about the same as ours. What will happen next? Will the tweaks be terminated or just reined in? By the time this is in print, we might have the answer.

Other audio and video publications.

Only the *Journal of the Audio Engineering Society, Stereo Review,* and *The Audio Critic* have so far made an unequivocal stand on double-blind listening comparisons at matched levels. All other audio and video publications are either fence-sitters or avoiders of the subject. (I'm not counting independent columnists like Larry Klein and Bob Pease.) For example, *The Sensible Sound,* a pleasant little audio journal on the whole, lists highly credible contributors like Tom Nousaine and Rich Modafferi (our new RF expert) on the masthead, but also prints gushing descriptions by lightweight reviewers of the nonexistent sonic personalities of various amplifiers and preamps. (Again, "equal time" for truth and nonsense.) Or take the superglossy, somewhat trade-flavored *Home Theater Technology,* whose number two executive is an avowed admirer of *The Audio Critic.* That doesn't prevent them from having Corey Greenberg as their technical editor, the same gonzo reviewer who learned at *Stereophile* how to talk the talk without having to walk the walk—how to write about big differences in sound between, say, two AV preamps, without double-blind listening tests, without measurements, without any kind of accountability. Check it out, Beavis—this preamp is cool, the other one sucks.

I could have picked other examples, in audio and video, current or recently defunct, but they all have basically the same approach: plug it in, listen casually, write whatever pops into your head, pretend you never heard of more accurate methods of testing.

The New York Times.

The Western world's newspaper of record, right? All the news that's fit to print, right? Well, when it comes to audio, the *Times* has not taken the scientific high ground. Far from it. Hans Fantel used to do their Sunday audio column and was not without some undisciplined dilettantish tendencies, but their new man, Lawrence B. Johnson, is a major disaster. He is a *Stereophile* editor, for crying out loud! He edits the *Stereophile Guide to Home Theater,* and his *Times* pieces fit perfectly the track gauge of the Atkinson, Archibald, and Santa Fe. Analysis of nonexistent sonic differences between electronic components, self-indulgent subjectivism, inaccurate technical explanations, assorted audiophile myths, the whole bit. A postscript by Larry Archibald is all that's missing. (I have even heard facetious/paranoid suggestions to the effect that Johnson is a mole, controlled by Archibald, cleverly infiltrating the U.S. daily press.)

It is difficult for me to believe that nobody at the almighty *Times* is aware of the schism I discussed at the beginning of this column—the schism between science and tweako cultism in audio—but that must be the case because no serious newspaper, especially not one with a superb science section like the *Times,* would knowingly side with the tweako faction and choose, of all possible choices, Lawrence B. Johnson.

Hey, hey, LB J, how many myths did you shill today?

Business Week.

At least one of the senior editors at this McGraw-Hill publication needs a new Rolodex. He is the editor that correspondent Tim Smart reports to. The Rolodex should have names like Stanley Lipshitz, Dick Greiner, Floyd Toole, Mark Davis, David Clark, Bob Adams, etc., on it. Why? Because Tim Smart, the magazine's usual reporter on audio matters, obviously talks only to tweaks and reads only tweako publications. He believes and tries to disseminate the untutored agenda of his sources (see Issues No. 16 and 22) and needs some serious factual correction from headquarters. The authorities on the suggested Rolodex could impart to his boss the scientifically correct information that nobody at McGraw-Hill appears to know how to get.

Question: Is it possible that *Business Week* is just as shaky on subjects I know less about?

Forbes magazine.

Ditto. They seem to be doing the same thing as *Business Week* (maybe not as often) and need the same Rolodex. It occurs to me that the two business magazines may be calling each other for audio information. Yikes!

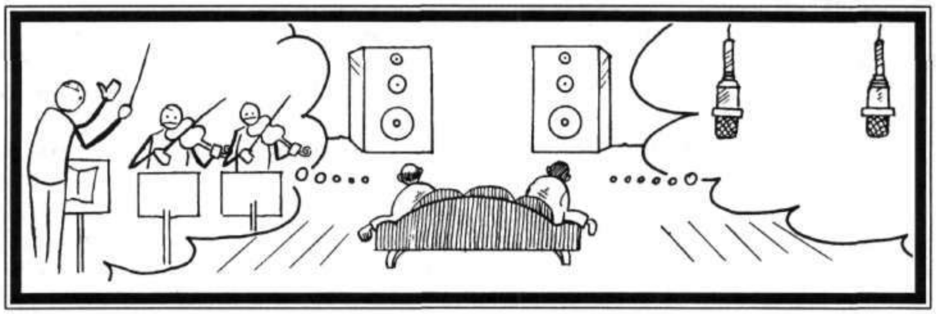
Letters to the Editor (continued from page 4)

MG-1.5/QR, whereas our small group here didn't think the speaker was even close to being recommendable to our readers. Superb transparency? Not to my ears or those of my associates. Indeed, I spent more time on this speaker trying to find a correlation between the sound and the measurements than I usually do when we didn't like the sound in the first place—and you don't think I did enough! Dr. Rich liked the sound of the speaker even less than I did; he impatiently dismissed the few small virtues I had found in it—and he goes to at least one live concert a week, sometimes two.

Maybe if you went to hear the Minnesota Orchestra once a week...

—Ed.

Recorded Music



Editor's Note: David Ranada was so late with his reviews for this issue (late even for me, the grand master of the late game) that I felt compelled, while waiting, to come up with my usual quota of capsule reviews, in the usual combination of new and not-so-new releases. I keep repeating that this is not really the way I want to run this column, that we need a degreed musicologist here (which David is), that we also need credentialed nonclassical reviewers, but my protests keep turning into "famous last words." Anyway, our readers appear to like my slight reviews more than I do. Hey...

Mostly Wagner and Mahler: CD Releases of Importance

By David Ranada
Contributing Editor at Large

Malcolm Arnold: *A Grand, Grand Overture; Concerto for 2 Pianos (3 Hands); Carnival of Animals; Symphony No. 2. Royal Philharmonic Orchestra, Vernon Handley, conductor.* ***Conifer*** *75605 51240 2.*

With instrumentation including full orchestra, pipe organ, three vacuum cleaners, a floor polisher and four rifles, you can't fail to be entertained by Malcolm Arnold's *A Grand, Grand Overture,* originally written for the one of the famous Hoffnung comedic concert music extravaganzas. Also Hoffnung-inspired is the *Carnival of Animals,* containing a movement (Chiroptera) inspired by bats, in which the orchestra apparently plays like mad, but silently, for about half a minute. (Of course, we'll have to wait for an HDCD recording of the work before we'll be able to "hear" the immense amounts of ultrasonic energy such batlike activity will generate!) The 1953 Symphony is most immediately fascinating for its occasional pre-echoes of textures, instrumentation, and chord progressions of music for the original *Star Trek* TV series. Engineering is by the ever-reliable Trigg vi Tryggvason, and the recording locale is the All Saints' Church, Petersham, Surrey, which probably accounts for the presence of the magnificent pipe organ sound at the parodistically extended ending of the *Overture.* If only all recordings of Strauss's *Also sprach Zarathustra* were so organ-ically blessed.

* * *

Richard Wagner: *Tristan und Isolde. Berliner Philharmoniker, Daniel Barenboim, conductor.* ***Teldec*** *4509-94568-2.*
Richard Wagner: *Die Meistersinger von Nürnberg. Bavarian Staatsorchester, Wolfgang Sawallisch, conductor.* ***EMI*** *CDCD 55142.*

Tristan und Isolde **should be** an easy opera to record. There are only a handful of main characters, no passages in which more than two of them sing simultaneously, and only a few, noncomplex passages for chorus in the first act. Yet there have been sonic problems with every single commercial stereo recording of the work, and the new Teldec is disappointingly no exception.

At least the orchestral pickup is magnificent. That you can seemingly hear everything in the score is a tribute as much to the Teldec engineers working in the Berlin Philharmonie as it is to Barenboim's skill in balancing orchestral sections. The basic sound of the orchestra is also magnificent, with a firm low end to the brass and a supremely smooth and blended string sound, probably the best Tristan playing on disc.

This recording's problem is with the voices, most of which are only barely able to get through Wagner's difficult writing without totally breaking down. Even the crucial secondary roles such as Brangäne and Kurnewal, which have been reliably cast on previous recordings, are here given to singers with basically unattractive instruments.

The engineering serves none of them well. The soloists are miked in such a way that there is very little "leakage" of the voice into the orchestral pickup. While you can hear a voice reverbing around the hall when the orchestral textures are thin and the dynamic scale is toward the quiet side, as textures thicken and the volume rises, the singers sound more and more as if they were in isolated sound booths, even though they weren't. As a result, the "size" of the voice, which is gauged by its relationship to the orchestral sound, is artificially reduced, especially if you play the recording too softly. While you never lose the words beneath the orchestra, the final result sounds faked, as if these voices needed electronic assistance to be heard at all. Besides, in every live performance of the work I've heard, there are passages when the orchestra does swallow the voices. If it weren't for the singing, this production would probably be the most recommendable of the stereo *Tristans.*

Beside the inadequacies of the vocal sound, the fadeout at the end of the first disc and overlapping fadein at the beginning of the second (the start of Act I, Scene V) is only a minor irritation. A musically appropriate and dramatically less distracting fade point occurs only a few moments earlier, on a solo timpani roll. (Disc 2, containing Act I, Scene V complete, is by far the most successful single passage in the performance, mainly because the orchestral textures are thin enough, most of the time, for the voices to sound naturally arrayed against them.) The translation in the libretto contains what seems to be virtually the standard version by Lionel Salter (it's included with the Polygram-label recordings by Solti, Bernstein, Kleiber, and Böhm). While accurate, it leaves out much of Wagner's scene-setting stage directions, turning the whole enterprise even more into an oratorio.

EMI at least has an excuse for the multiple engineering inadequacies of its new *Meistersinger* recording: the work is Wagner's most complex. The riot scene in Act II, for example, has all the major characters singing different simultaneous lines, along with a subdivided chorus and full orchestra playing multiple leitmotivs in the complex counterpoint that distinguishes *Meistersinger* textures from all of Wagner's other music.

Those textures are not clarified particularly well by EMI's recording, which among other things often has part of the first violin section leaking into the right channel (obviously a violin microphone panned to the wrong side in the mixer). While this fault is most obvious on headphones while following a score, other sonic problems are clear even over speakers. For instance, at the end of Act II, the Night Watchman disappears to the left while the horn he is supposed to be playing disappears to the right. And I leave it for the listener to find the weird two-second multiple-take overlap that was allowed to pass. These are only a few examples of the sloppy engineering characterizing this release.

The basic orchestral image is that of a multimike production, with certain orchestral sections becoming unpleasantly mono-sounding at times.

The bass drum during the entrances of the trade guilds in Act III—Wagner's only use of the instrument in his mature operas—is anemic. It would normally produce a pleasant surprise in an opera house. On the whole, the engineering in this production is a far cry from the stunning clarity EMI achieved in their Bavarian recording of Humperdinck's *Hansel und Gretel* **(EMI** CDCB 54022), easily the best-sounding recording of that Wagnerian work. It is even inferior, in basic orchestral and voice pickup, though not in distortion or noise, to the older Kubelik performance **(Myto** MCD 925.69), a recording that reportedly was to be released on Deutsche Grammophon.

Die Meistersinger still needs a clearly superior stereo recording. The sonic inadequacies of this EMI production balance out its many vocal felicities and conductor Sawallisch's superb pacing, to add up to a rendition equal to but not better than Solti's **(London** 417 497-2). The upcoming second Solti *Meistersinger,* to be recorded live in Chicago's Orchestra Hall, will probably not fill the bill, to go by the Solti live-recorded rendition of Verdi's *Otello* **(London** 433 669-2), a performance notable for its lack of dramatic tension. I hope to be proven wrong.

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Gustav Mahler: *Symphony No. 9. Berliner Philharmoniker, Herbert von Karajan, conductor.* ***Deutsche Grammophon 439 024-2.***

Gustav Mahler: *Symphony No. 9. Columbia Symphony Orchestra, Bruno Walter, conductor.* ***Sony Classical*** *SM2K 64452.*

Gustav Mahler: *Symphony No. 9. Polish National Radio Symphony Orchestra, Michael Halász, conductor.* ***Naxos*** *8.550535/36.*

Readers of my previous reviews will know of my obsession with orchestral seating plans, specifically the use of the standard turn-of-the-century string arrangement plan—first violins and cellos left, second violins and violas right—with music written for it, which basically means all orchestral music written between around 1850 and 1925. Among all the orchestral pieces of this period, which make up the core of the symphonic repertory, in no other work is the musical importance of this arrangement more significant than in Mahler's 9th symphony.

Throughout his career Mahler granted the second violins an unusual amount of independence; they had previously been an orchestral section that had mostly subsidiary or backup material to play. In Mahler not only do the "seconds" perform their standard task of reinforcing the "firsts," but he often gives them prominent autonomous melodic lines. Mahler also bases many of his special effects, such as sudden transferals of background textures or melodies from side to side, on the seconds' placement opposite the firsts. It is one of the joys of those who can read an orchestral score to be able to follow such devices, even with monaural recordings.

The 9th symphony is outstanding even among Mahler's works for having second-violin lines of absolutely critical prominence. You need only play the first couple of minutes of the first two movements to realize that the second violins are carrying the main melodic burden, the firsts being silent until quite a bit later each time. Throughout the rest of the piece you'll often find the two violin sections' melodic lines continuously intertwining, so much so that without a left/right split of the two sections you'll have a hard time deciding which notes belong to which line. Such activity continues until the final pages of the score, where the melodic lines disintegrate into single notes traded across the width of the sonic stage. The highest note of the final chord belongs to the seconds, the firsts are silent. Without following a score you can hear these things only if you are listening to a recording having a separated left/right violin arrangement, the sonic differences between the two sections in an all-violins-left recording being usually nonexistent.

Many conductors seem to have recognized the importance of separated left/right violins for Mahler's 9th symphony, if not his others. The 9th is probably the Mahler symphony that has been most often recorded with left/right violins. The commercial stereo recordings by Abbado (DG), Morris (IMP Classics), Kubelik (DG), and Klemperer (EMI) are the ones I know about; there may be others. A notable new one is that conducted by Michael Halász on Naxos, the only digitally recorded budget-priced Mahler 9th. He uses what might be called a "compromise" orchestral layout: cellos remain on the right, but the violins are split and the violas are on the left.

While not absolutely authentic historically, the arrangement works extremely well here, aided by Naxos's clear, well-balanced and lifelike sound, perhaps the best-sounding, least overproduced Mahler 9th since Inbal's on Denon—and at a budget price, no less! The sound lacks only a little firmness in the bottom two octaves (20 to 80 Hz) that would have given the double basses and the rare but crucial bass drum passages more solidity.

The performance itself is excellently paced, especially the last movement, which has too frequently been stretched to nearly a half hour's duration to the detriment of the movement's dramatic proportions (it's 24:23 here). Unfortunately, Halász makes very little of Mahler's copious, even obsessive, markings for short-term dynamics (loudness variations within a single note) and articulation (defining the attack and release of each note). Without close, even exaggerated, observance of these markings, the music will sound grand but bland, as it does with Halász and does *not* under such noted Mahlerians as Tennstedt and Bernstein. Still, the Naxos release is the least expensive Mahler 9th with left/right violins and as such is a low-risk investment for any serious Mahlerphile.

Another Mahlerphile must-have is Sony's latest remastering of Bruno Walter's classic stereo recording of Mahler's 9th with the Columbia Symphony Orchestra (here, a pickup ensemble made up of Los Angeles musicians). This is not so much for the performance, which is only a little less bland than Halász's and as such quite different from Walter's gripping 1938 live performance with the Vienna Philharmonic (on **EMI** CDH 63029), the first recording of the piece and one that Walter always considered unsatisfactory. Nor is it for the sound, which was superb for its era (1961)—and much less harsh here as well as slightly lower in noise than in its previous Columbia CD incarnation (averaged spectra show a rolled-off treble response in the newer release compared with the old one)—but no match for any of the more modern digital recordings, much less Denon's or Naxos's.

The reason why every Mahler lover should have this album is that there are two "bonus" tracks on the first disc. One is an interview with Walter on his relationship with Mahler as pupil and musical protege (Walter conducted the world premiere of Mahler's 9th in 1912, the year after the composer's death). The other bonus track is a fascinating glimpse into Walter's rehearsal technique as he explores the 9th with musicians who had probably never played the piece before. You'll hear Walter memorably explaining some of Mahler's orchestration tricks. The program notes by Andreas Kluge are unusually clearheaded and nonidolatrous on the subject of Walter's relationship to Mahler and his music.

Walter's 9th is part of a mostly complete set of Sony remasterings of his Columbia recordings, of which the discs of Mahler's music are musically the most important. After the 9th, I'd rank in importance the recording of *Das Lied von der Erde* (still one of the better sung versions) and the 1st Symphony. Less critical are the poorly recorded 2nd Symphony and the mono-era 4th and 5th Symphonies. In the 5th the remasterers have let stand an irritatingly large amount of disc noise from their original source materials, putting sonic values above musical ones.

Karajan's live DG recording with the Berlin Philharmonic of the Mahler 9th has been considered one of the great ones since its first release (like Bernstein's with the same orchestra on the same label). It has also been refurbished recently as part of Deutsche Grammophon's Karajan Gold series. The remix job has produced a closer view of the orchestra, with far less hall reverberation than before. As a result, the short-term dynamic inflections that Halász missed emerge even more vividly under Karajan. There also seems to be more bass firmness in the new mix, though long-term spectral averaging shows that the overall balance has not been changed by equalization. Passages containing bass-drum rolls send the waveform into clipping more often than in the first release, though the clipping is inaudible as such. If you have a copy of the earlier release, the changes are not large enough to warrant buying the remastering. But the performance, inauthentic seating plan and all, remains magnificent—and this from a reviewer who, in general, hasn't liked Karajan's interpretive approach. The Karajan Gold remasterings do raise some interesting "ethical" issues, however. Can Karajan truly be said to be the conductor of this performance when the resulting orchestral balances were not heard nor approved by him? •

first-rate; the conductor is quite famous locally and obviously excellent. As for the sound, the Great Hall of the Moscow Conservatory has superior acoustics, the dynamic range of the recording is outstanding, low-level detail is particularly fine, and an overall naturalness is very much in evidence. Gene Pope claims an utter breakthrough in audio quality; I'm inclined to wait for more PopeMusic releases before seconding that.

Reference Recordings

This is the only label I'm aware of that has completely bought into HDCD, the digital audio enhancement technology claiming to correct for the "shortcomings" of ordinary linear PCM without disclosing how it's done. (See also my review of the EAD DSP-7000 Series III in this issue. That's the HDCD-equipped processor I used to audition this recording.)

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***George W. Chadwick:** Symphonic Sketches (Jubilee, Noël, Hobgoblin, A Vagrom Ballad); Melpomene Overture; Tarn O'Shanter (symphonic poem). Czech State Philharmonic, Jose Serebrier, conductor. RR-64CD (1995).*

George Whitefield Chadwick (1854-1931) is a seriously neglected but far from negligible American composer in the general stylistic mold of Brahms and Dvorak. These are highly listenable pieces, beautifully played and conducted here. Good music, good orchestra, good conductor—those are the reasons why this CD is worth having. As the first HDCD release, ever, of symphonic music with strings, it is on the other hand very disappointing. The HDCD-decoded sound is distant, hollow, poorly balanced, blunted on top, and strangely restricted in dynamics—far from natural. One is tempted to use the word *processed.* What could be the weak link? Keith Johnson, although not as experienced in recording the full orchestra as, say, John Eargle or Jack Renner, has certainly made some superb symphonic recordings (e.g., the Arnold overtures on RR-48CD), so why not in this instance? Maybe he was totally unfamiliar with the hall (Stadion Hall in the Czech city of Brno) and was forced to work under pressure. But maybe it's that hush-hush, tweako HDCD process? Maybe it can't sound natural? I'll reserve judgment on that—but not forever.

Telarc

Slowly Telarc is running out of standard repertory pieces to record in great digital sound and may have to turn to more esoteric music to keep the label going. Or will it be Brahms and Tchaikovsky all over again in 20-bit versions?

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***W. A. Mozart:** Le nozze di Figaro. Alastair Miles, Figaro; Nuccia Focile, Susanna; Alessandro Corbelli, Count Almaviva; Carol Vaness, Countess Almaviva; Susanne Mentzer, Cherubino. Scottish Chamber Orchestra & Chorus, Sir Charles Mackerras, conductor. CD-80388 (3 CDs, 1994).*

Mackerras seems to be intent on repeating the mid-1930s "miracle of Glyndebourne" and record the digital era's model version of each major Mozart opera. This is No. 3, after *Die Zauberflöte* and *Così fan tutte.* He has the musical culture, the conductorial ear, and the orchestra to go all the way; I don't think he has the voices. This *Figaro* is effervescent, authoritative, complete, and wonderful in 17 different ways, but the singing in it is musical and spirited rather than beautiful. I was spoiled in my youth *Figaro-Wist* (Siepi as Figaro, Steber as the Countess, etc.) and am unable to be thrilled as I listen to this. Too bad, because the next more excitingly sung version will undoubtedly lack the musicianship of this one. I'll leave the details to the more scholarly reviewers. Jack Renner's 20-bit recording is all one could ask for (same venue, same equipment as in the *Cosi* of 1993).

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***Gioacchino Rossini:** "Overtures." Guillaume Tell, La gazza ladra, L'italiana in Algeri, Semiramide, La scala di seta, Tancredi, II barbiere di Siviglia. Atlanta Symphony Orchestra, Yoel Levi, conductor. CD-80334 (1992-93).*

Recorded in two sessions almost a year and a half apart, this collection was then delayed more than a year before being released. I don't know why Telarc wasn't more excited about it because, in terms of balancing performance against sound quality, this is arguably the Rossini-overture CD of choice for the discriminating audiophile. The playing is crisp, buoyant, accurate; the surprise sforzandi are delightful; yet corniness is happily avoided at all times. The *Semiramide* overture receives one of the best performances known to me (shades of Toscanini!); the others are not quite as good but close. The Atlanta orchestra sounds like one of the Big Five here, especially the woodwinds, and Michael Bishop's stunning demo-quality recording gives them an almost unfair advantage over their compeers. Good stuff!

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***Arnold Schönberg:** Verklärte Nacht, Op. 4; Pelleas und Melisande, Op. 5. Atlanta Symphony Orchestra, Yoel Levi, conductor. CD-80372 (1993-94).*

This is the postromantic Schönberg, not the later dodecaphonist. That means the music is enjoyable, not just mandatorily admirable. Wagnerians like me eat it up, of course. The main course here is not the superb but smaller-scaled *Verklärte Nacht,* of which many better versions exist, but *Pelleas und Melisande,* which is scored for a huge post-Wagnerian orchestra and gives the excellent Atlanta forces, not to mention recording engineer Michael Bishop, the chance to produce some wonderful sounds. Not a masterpiece, not for everybody, but highly recommended if you dig it.

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***Franz Schubert:** Sonata in A Minor, D. 959; Moments Musicaux, Op. 94, D. 780. John O'Conor, piano. CD-80369(1993).*

The sonata is one of the posthumous three from Schubert's miraculous last summer (1828). It is sublime, but O'Conor's playing is too laid-back and monochrome (albeit fluent and polished) to project adequately the sprawling magnificence of the work. The lighter, more casual *Moments* fare better, and Jack Renner's B&K 4006 recording is exemplary, exactly as close-up as I like it.

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***Richard Strauss:** Salome's Dance; Suite from Der Rosenkavalier; Burleske, Op. 11; Festival Prelude for Organ & Orchestra, Op. 61. Cincinnati Symphony Orchestra, Jesús López-Cobos, conductor; Jeffrey Kahane, piano (in Burleske). CD-80371 (1994).*

Another 20-bit sonic blockbuster engineered by Michael Bishop. Only the *Rosenkavalier* music is Al Richard Strauss here, but everything the man ever composed is hugely enjoyable as sheer sound, and this CD indulges you with that near-guilty pleasure as few others. Want to show off your $50,000 system to a visitor? The lush strings, big bass, and open soundstage of this recording will do the job. The Cincinnati forces play better than I recall them in any previous effort, and López-Cobos conducts them like a dedicated Straussian. Kahane is suitably flashy in the *Burleske.* I loved every minute of the 65 on the disc.

Teldec

It's nice to observe that the supposedly ultra-commercial Warner empire is making recordings that often rival any audiophile boutique label in sound quality.

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***Aaron Copland:** Old American Songs (Orchestral Version); Down a Country Lane; Eight Poems of Emily Dickinson; Billy the Kid (Selection). Dawn Upshaw, soprano; Thomas Hampson, baritone. The Saint Paul Chamber Orchestra, Hugh Wolff, conductor. 9031-77310-2 (1992-93).*

Apple pie? This is as American as pumpkin pie at Thanksgiving! Two of America's finest singers perform one of America's finest composer's settings of some of the finest all-American texts. The Emily Dickinson poems are definitely the centerpiece here, but there's also my favorite nonsense song, "I Bought Me a Cat" ("My cat says fiddle eye fee"), and all sorts of other goodies. Great singing, good playing by the half-sized orchestra, very acceptable sound (but not Teldec's best).

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***Engelbert Humperdinck:** Hansel und Gretel. Jennifer Larmore; Ruth Ziesak; Hildegard Behrens; Bernd Weikl; Hanna Schwarz; Symphonieorchester des Bayerischen Rundfunks, Donald Runnicles, conductor. 4509-94549-2 (2 CDs, 1994).*

The Bavarian radio orchestra must be able to play this in their sleep because they recorded it only four and a half years earlier for EMI, in the same hall yet, with some overlaps in the cast and Jeffrey Tate conducting. This new version is a little lighter in touch, less "operatic," better balancing the childlike aspects of the work against the Wagnerian harmonies and orchestration. The recorded sound is also better, state-of-the-art I'd say; more transparent, freer on the dynamic peaks, firmer in the bass than the already excellent EMI. Of the versions I know—and I've been listening to this since the age of eight—this is as good as the best. Of course, to appreciate this music to the fullest extent, you must dig the whimsical quasi-Wagnerisms of the score.

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***Richard Wagner:** Tristan und Isolde. Siegfried Jerusalem, Tristan; Waltraud Meier, Isolde; Matti Salminen, König Marke; Falck Struckmann, Kurwenal; Marjana Lipovsek, Brangäne; Berliner Philharmoniker, Daniel Barenboim, conductor. 4509-94568-2 (4 CDs, 1994).*

I just want to add a few comments to David Ranada's longer review above. I agree with his overall view of the recording, but the singers are a little better than he avers. Meier is a musically very credible Isolde, if no Flagstad, and Jerusalem, likewise no Melchior, at least doesn't shout here as is his wont and sings intelligently. Barenboim's terrific conducting took me by surprise; he has advanced several pegs in my rankings.

Troy

This label is a division of Albany Records.

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***Jacques Ibert:** "Jacques Around the Clock" (Chamber Music for Flute). Sue Ann Kahn, flutist, et al. Troy 145 (1991-93).*

This collection is interesting for a number of reasons. These short, light, 1920s/'30s/French-modern pieces are unavailable elsewhere. Sue Ann Kahn is an excelent flutist who has the idiom down pat. Her collaborating colleagues are also fine musicians. The recording is by Max Wilcox in the American Academy of Arts and Letters, meaning it's lucid, natural, lovely-sounding. Most important of all, the music is sheer fun. Try it.