THE B.A.S. SPEAKER

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In This Issue

The June issue is devoted almost entirely to digital recording. It includes Scott Kent's meeting summary of the visit from Digital Recording Services, Inc. DRS provides digital mastering and editing services for record companies who wish to try out this new technology without investing thousands of dollars for new equipment. Many major labels have used this service, and we got to hear excerpts from these productions as well as some interesting stories about how things are done out there in the commercial world.

The rest of the issue comprises a review of the Sony PCM-1 digital adapter by Brad Meyer. This piece started out as a simple user's report and turned into a lengthy study of both analog and digital recording processes; it contains some illuminating lab tests of the two types of recording systems as well as the usual subjective impressions.

In order to expedite the release of the next three issues of the Speaker we have shifted the contents around a bit. There is accordingly no "In the Literature" in the June issue; there will be a double-length version in next month's Speaker, which if all goes as planned should reach you about a week to ten days after this one. The July issue will be devoted largely to a massive report by Peter Mitchell on the Chicago CES, and will be typed in an experimental format that may be easier to read than our traditional one.

For August we will return to the old format, and will be catching up on contributions from out-of-state members. Of these we have accumulated a fair number, although probably not quite enough to fill up both August and September. Our thanks go to those of you who have been sending us things in response to our many requests; now if we can get a few longer pieces for the back of the book we'll be in pretty good shape. But whatever we have on hand, we must get back on schedule, so September will be out soon even if it winds up being skinny.

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*Revox A77 Mk IV 1104 deck, "Kent" mods, timer control box, dust cover, $850; Revox A78 Mk II integrated amp, $300; Pioneer CT-F1000 three-head cassette deck, $325; Computer-type audio rack with latching front and rear doors, 19" mounts, $150; BASF C120 cassettes, fifteen for $40; Citizen calculator watch, full scientific functions, $150. Will trade for Yamaha C1 preamp or ADS time delay system. Jay, (617) 661-6611.

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Wanted

*Norelco full-range drivers: 12" (like AD 5200M or AD 5277M) and the best 8" (like AD 4877, etc.) in good condition. Bill Kalish, 565 Walnut Avenue, Redlands, CA 92373.

*Dahlquist DQ-10s with white grilles and oak trim, latest stands and mods; Futterman H-3aa (latest version); Stereo Record Guide (Ivan March, ed.), Vol. 8. (519) 945-8486 evenings, weekends, or before 8 AM EDT.

In Memoriam-Sheldon E. Feinstein

Sheldon Feinstein, President of Cizek Audio Systems, died suddenly and unexpectedly in July. Sheldon helped found Cizek in 1976, and although his original role was in the business end of the company, he contributed many ideas, relating to performance as well as marketability, to its products. While Cizek continues to function in his absence, he will be missed.

Turntable #28 Follow-Up

(Editor's Note: This communication comes from the owner of one of the turntables tested in the BAS turntable/arm/cartridge clinic. The unit in question is a Kenwood KD-550 integrated turntable/arm with a Stanton 881S cartridge.)

Prior to the turntable clinic, I had guessed at the existence of the sharp, potentially troublesome arm resonance illustrated on page 22 of the February 1980 Speaker (Vol. 8, No. 5) when a rather warped pressing of a favorite recording (Music from France for Oboe and Orchestra, RCA Red Seal LSC-2945) began to cause mistracking. Since the clinic, I have removed the Stanton brush, installed a Denon low mass headshell, and installed a Disctraker (with all-important plastic nuts and bolts). No more mistracking, but the resulting front end mass reduction on the clunky Kenwood arm requires a counterweight position so close to the arm pivot assembly that it interferes with arm lifting. Since I have not yet "drilled out" the counterweight, I usually operate with the Disctraker installed for mass, but not contacting the record. When using the Disctraker for warpextremis, I find that it operates best when the lower available cylinder travel is about 1/3 of the cylinder length (i.e., set the piston below the midpoint). I have also found that, to my brass ears, my system is significantly "tightened up" if I place a heavy book (e.g., McClane's New Standard Fishing Encyclopedia) on top of the closed KD-550 dust cover. Of course, the mass increase moves the suspension resonance downward, as does installing A1 Foster selected supple springs. The nice feature of the book treatment is the fact that it also damps dust cover resonance, and thus reduces any air transmitted feedback at frequencies which would otherwise excite the cover. With the stylus resting on a record and with the volume at a high level, a light tap on the dust cover provides a clear demonstration of the potential for such excitation. -- Paul Ruenzel (Massachusetts)

Further Hi-Fi Choices

My summary, entitled "Hi-Fi Choice(s), 1980," in the April 1980 Speaker (Volume 8, Number 7, pages 5 and 6) stands in need of a little correction and some updating. To be pedantically accurate, Hi-Fi Choice No. 13 is actually copyright 1979. More importantly, when ye editor deleted a conjunction from the first sentence of my penultimate paragraph, he had me say that Hi-Fi Choice tested headphones on a Neumann KU80 dummy head with a Bruel and Kjaer 4153 artificial ear in some kind of simultaneous combination. But a dummy head has two ears already l These are simply two separate tests carried out in the attempt to provide additional objective data, which attempt proved successful in the high degree of corroboration between the two tests as shown in those data.

As for the updating, Hi-Fi Choice No. 18, "Turntables and Tonearms 3," copyright 1980, has arrived. The methods of investigation I outlined in my April summary have changed little, involving mainly the remounting of the accelerometer in order to increase its sensitivity to torsional resonances, and increased reliance on the Mission 773 pickup cartridge for subjective assessment.

In a field of eighty-five turntables, five now rank in the top group: the Ariston RD 11S, Dunlop System-dek, Linn Sondek LP 12, Strathclyde STD 305S, and Thorens TD 160S. This last item differs from the 160BC in having its subchassis acoustically damped by a bituminous laminate, its suspension improved by removal of the foam cores, its arm mount reinforced, its plinth (base) made of heavier wood composition with a massive bottom cover, and its standard ribbed rubber mat replaced with a smooth one.

Some revision of the rating scale for separate pickup arms was done to accommodate improvements here without running out of superlatives. The top category now embraces seven in a field of twenty-seven: the ADC LMF-1, Grace G707, Linn Ittok LV11 (striking treble clarity), Mission 774, SME 3009III/IIIS, Syrinx PU2 (high horizontal pivot axis apt to degrade tracking), and Technics EPA-500 (slightly "soft"). -- Jack Reed (Illinois)

More Raves for the Ruby

What helps to make our lives so pleasant at times are the little surprises the world brings us: a new puppy or kitten for a child, a diamond engagement ring from your fiance, etc. In audio, a few select manufacturers occasionally bring us glimmers of joy also: the Keith Monks' Record Cleaning Machine, Conrad-Johnson, Audio Research and Beveridge tube equipment, Quad, Magnepan, Infinity and Beveridge speaker systems, Cotter, Linn and Oracle turntables, etc. Most of this "sunshine" comes to our listening rooms through massive bank loans. What you get, though, when you're finished (if all the equipment is matched/interfaced correctly, and is set up very properly in the listening area) is a major step closer to the sound of Live Music.

Here comes Dynavector with a new "cheapie" contender -- the Karat/Ruby Cartridge. How in the world can a $275.00 item compete with the likes of Koetsu, Linn-Asak, Levinson, et al? Well, it appears, with some caveats, that the answer is -- superbly.

First, we ought to discuss what this cartridge is not. (A) It isn't for those who rough-handle their disc playing equipment. Like most moving coils, the stylus isn't user replaceable and is <u>very fragile.</u> (B) The Ruby is not for those of you who can't or won't take the time and effort to set it up properly in your tonearm, so as to extract the last iota of performance that it is capable of providing. (C) This is probably not a good choice for those of you who believe that a cartridge purchase is much like buying electronics (amplifiers, preamps, etc.); that is to say, it very likely isn't going to last three years. (Editor's Note: There is actually some reason to believe that the Karat/Ruby will sound good for <u>longer</u> than some other cartridges; see the following review.) (D) The Ruby won't make you happy if you're one of those who don't understand or won't take the time to choose a compatible tonearm. The Ruby likes a low-to-medium-mass tonearm. Some of the acceptable choices are: (1) the Grace 707 MKII or new 747; (2) the Mayware Formula IV; (3) Magnepan's new uni-pivot arm, the Unitrac I; (4) the Infinity Black Widow II -- so it is reported to us by reliable sources, although we're amazed considering its very loose bearings; (5) the Pioneer Series 20 PA-1000 (discontinued), if you can set the VTA correctly -- the arm tube when mounted on most tables cannot be lowered far enough. Here a <u>glass</u> platter mat, like the new Marcof, placed on top of your platter will probably help alleviate the problem. More than likely there are other candidates, but these are the ones we've had experience with. We have not been able to use one of our favorite arms, the superbly engineered Lustre GST-801 (which has user adjustable VTA while the record is playing), but we believe that it very possibly will produce excellent results. However, with this relatively light cartridge, with its medium to medium-high compliance, a large, massive arm like the Fidelity Research FR-66S would not be a wise choice.

So what's the good news? When properly installed and aligned in the correct tonearm, and terminated into 35 ohms, this cartridge is capable of extracting the "nth" degree of detail from your record collection. What this means is that you can actually sit back, listen and "re-discover" all of your records. Subtle and heretofore unheard nuances will now be reproduced by the Ruby. At times, it can send shivers up your spine. For example, on the "Rosie O'Grady Goodtime Jazz Band" Direct-Disc Labs record, you can actually hear mumblings from either the musicians or the recording crew between selections for the first time. But the best news is what the Karat/Ruby does with vocals, especially female. Voices are extremely difficult to reproduce naturally. This the Ruby does quite well. Voices, more often than not, tend to be diffuse and rather non-specific in location. The Ruby renders them focused and highly detailed. It's like the cartridge grabs the performers with an iron fist and makes them stay absolutely stationary. Distortion with the Ruby is vanishingly low. Dynavector has managed to lower it by (what appears to be) a whole order of magnitude. You just keep wanting to turn up your system louder and louder, the sound is so clean.

We have had success using two different step-up devices for the Karat/Ruby: the Marcof pre-preamp and the Audio Interface Transformer. The Marcof PPA- 1 (new version, regular output, in the black case -- not the older version in the blue box, nor the "higher" output model, generally used with the Fidelity Research FR1 MKIII-F) at $125.00 does an excellent job, with good bottom-end response. The upper midrange and high-end could be better, and it's still a little too noisy, especially on wide dynamic range recordings. The best luck we've had is with the Audio Interface Transformer. We understand that it may have been engineered with the Karat series in mind. This little $300.00 black box is ever so much smoother and sweeter with the Ruby than the Marcof, and much quieter also. For the money, though, the Marcof is an acceptable choice. One additional unit we haven't tried, but which has been reported to be excellent, is Dynavector's own Silver Transformer, Model DV/6A.

It should be noted that while we had been hearing scattered "rave" comments about the Karat/Ruby, it took the review in The Absolute Sound (Issue 17) for us to gamble on its purchase. TAS has become more and more accurate in their assessments of equipment and discs as the years go on. In this case, we find their review of the Ruby to border on understatement of its virtues. By the way, it's been said that Harry Pearson took great pride in this gigantic cartridge testing project -- and he should be very proud of the results.

It may be difficult to find a Dynavector dealer who has the Ruby in stock (my nearest dealer is the excellent House of Stereo in Jacksonville, Florida, headed by Bill Gibson and his well-trained staff). Dynavector, at least as we view the situation, had not built a large, strong dealer network. The DV-505 tonearm did not add to the company's credit in most quarters although some say it is an excellent performer with the Karat series. It should be noted here that Dynavector is now engineering two new tonearms for the Karats. However, the emergence of the Karat/Ruby and Karat/Diamond has placed horrendous demands on the U.S. importer/distributor, Mark Schifter of Dynavector Systems. These cartridges are almost hand-built (when you see how tiny the solid ruby cantilever is, you'll understand), and production in Japan can in no way keep up with world-wide demand. You'll likely not see this cartridge offered like a Shure V15 Type IV at "$150.00 List -- But our mail-order special is only $81.25!" And rightfully so, because you'll want a dealer who will stand behind his products, and, in this case, properly set up the cartridge for you. It just can't be stressed enough how important it is that all the parameters of cartridge mounting be followed to the letter, and then optimized for your particular set-up. Overhang, alignment, VTA, anti-skating, etc. must be very close to perfect for you to be able to enjoy this very musical device to the fullest.

One last note: you must be very careful with stylus cleaning. The stylus-to-cantilever mounting does not allow you to use your typical stylus cleaning mechanisms (i.e., the back of your Discwasher Record Cleaning Brush, alcohol, your finger-tip, D/3 fluid, Q-tips, etc.). Even the Discwasher Stylus Cleaning Brush is probably too rough for the Dynavector. It is recommended that you use a dry, very small artist's paint brush, or a soft stylus brush like the one supplied with Shure V15 cartridges. If you need additional cleaning, you might lightly moisten your brush in tap water.

So, if you're into measuring the RIAA equalization of preamps, or believe that nothing tracks better than a Shure V15 cartridge, the Karat/Ruby probably isn't the one for you. But if you're seriously into music, listening to and enjoying it, then you might want to give the Ruby a try. Dynavector has given all of us the opportunity at a relatively reasonable price.

-- Mike Lulejian (Georgia)

For the past seven months I have been using the Dynavector Karat (Ruby) phono cartridge. It has some very captivating traits and breaks new ground in a few respects.

I obtained an early sample from Mr. Hamburger of Audioworks in Harrisburg, Pennsylvania. At the time I had not heard anything about the cartridge, but Mr. Hamburger said it was pretty special. Since then the unit has received a rave review in the latest issue of The Absolute Sound. They place it in their highest category and venture that it is perhaps #1 in information retrieval ability, although they maintain that it is important to have a correctly matching load impedance in the transformer or head amp for this moving coil device. (Yes, they say that even head amps should probably have an impedance matching characteristic, which is a new one on me.)

The Dynavector Karat weighs 5.3 grams and tracks at about 1.5 grams. Since its compliance is somewhat on the low side, it might be helpful when using an ultra-light arm to add some mass to the arm to help bring the cartridge-arm resonance into the proper range. This can be pinpointed with the new Ortofon test record designed for the purpose.

I have always sought a cartridge that would bring out greater detail from the record. Call it more transparent, less veiled, better definition, more accurate or whatever is the current terminology. This cartridge makes one wonder what more there could possibly be in the way of musical information in the record grooves. In fact, I am left with the strong feeling that typical American-made records are better than we thought.

Old records show up with nuances that were not discernible before. I have some recent records that had sounded excellent except for a rare, small lapse of blurred, fuzzy distortion during an extremely difficult moment in the sound. Since I had been using a fine cartridge before, the Coral 777 EX, which is the Japanese equivalent of the GAS Sleeping Beauty Shibata, I assumed the distortion was a problem of the record or the recording microphone. However on playing with the Dynavector, it turned out there was clear music throughout after all.

My associated equipment comprises a very high-resolution system. The speakers are stacked Quads with Pyramid ribbon tweeters and the Janis W-1 subwoofer. The preamp is the new Theta (no-negative-feedback) unit and the cartridge is mounted in Dean Slindee's Rabco mod Audioetc arm and played through the Verion transformer with PP strapping, which is within a few ohms of the ideal impedance match. According to The Absolute Sound, Dynavector's own silver-wire transformer would be better.

The cantilever/stylus assembly is the most distinctive feature of the cartridge. The cantilever is formed from a single piece of artificial ruby. The diamond stylus is neatly fitted to the ruby through a hole drilled with laser technology.

When held up to the light and observed with magnification this assembly is the most beautiful piece of engineering I have ever seen in a cartridge. It is an absolutely clean piece of geometry without the usual globs of adhesive and other odd protuberances you see when looking at most stylus assemblies. The whole thing is also translucent.

The cantilever itself is very short and is not damped at all. The idea of a non-metallic cantilever without damping appeals to me very much because of our hot, humid tropical climate here in Puerto Rico. This can cause some damping materials to deteriorate after only a few months. Also we have some corrosive salt spray in the air from the ocean. All other cantilevers I have seen, even ones made of aluminum, develop quite a build-up of corrosion-related crud. The usual treatment that keeps the stylus clean does not do much for the cantilever, and although I have no idea what effect this might have on performance, it has always made me nervous to see tiny things growing on my cartridge's cantilever. So far nothing has grown on my Dynavector phonocartridge but beautiful music. -- Lohr H. Gonzalez (Puerto Rico)

(Editor's Note: Some of you may have noticed that the previous reviewers disagree on the compliance of the Karat/Ruby, and therefore on what tonearms will work best with it. Local member (and Dynavector dealer) Dick Glidewell supplied us with the information that both are right, sort of. The compliance of the Karat/Ruby is about $15x10^{-6}$ cm/dyne, which is stiffer than the most compliant moving magnet cartridges (which range up to 30 or so) and softer than many moving coil units, which until recently tended to fall below 10. A medium-mass tonearm is probably ideal, but Dick has seen the cartridge track well in very low mass arms.

As for the alleged fragility of the stylus/cantilever mounting, some early samples of the cartridge had a cantilever design problem, which has since been fixed; otherwise there seems to be no reason to take precautions beyond those suggested by the unit's non-user-replaceable stylus and $275 price.)

May BAS Meeting

May's BAS meeting at GTE laboratories, Waltham, began with Peter Mitchell mentioning various items for sale, after which he turned the meeting over to Dick Lewis of Advertising Assistance in Weston, Massachusetts. Dick, whose firm handles advertising and PR for our guests of the evening, introduced the first meeting feature, Digital Recording Systems Co. of Elkins Park, Pennsylvania. DRS is engaged in two track digital recording, mixing, editing and mastering. All services are provided on location anywhere in the world and mixing or editing can also be accomplished in their Elkins Park studios. They use the Sony PCM-1600 digital encoder/decoder system, with two Sony BVU-200 U-matic video cassette machines. Mark Levinson and BKM Associates electronics are used to interface the Sony system with the analog (outside) world. Available for location recording are three Levinson-improved B&K 4133 one-half-inch microphones. The Elkins Park studio contains a complete Levinson HQD playback/monitor system.

Lewis then introduced Terry Tobias, president, and Peter Jensen, technical engineer, of DRS, and also John Sullivan of BTX, Inc. BTX provided a controller for synchronizing two video machines to allow an audio/visual presentation of a section of this past season's Metropolitan Opera production of "Otello" (not Otala -- that was last season's BAS production -- SK). Additional equipment provided for the meeting demonstration included eight Cambridge 310 loudspeakers, a Dunlap-Clarke Dreadnaught 1000 amplifier, and a Sontec parametric equalizer. (This latter item was to "equalize" the room despite our April meeting's guest who showed this to be impossible. As readers of the Speaker know, the BAS is an organization of skeptics, and it is as difficult to convince us that something is impossible as to convince us that anything is correct.)

Terry Tobias spoke briefly of the industry's need for an independent digital recording service. In any large, established industry, companies are reluctant to invest headily in a new technology, particularly in an area where no standards exist. Having a system to try out before making an investment is helpful both to convince skeptics and to de-bug digital hardware in real-life situations. The opportunity to use and evaluate different systems will help in the formulation of proper standards. An independent team, separate from the hardware manufacturer, can serve as a more objective proponent of digital recording than the company who would profit directly from the sale of the equipment. As well, until the hardware cost drops significantly, investment in digital by smaller record companies who haven't the workload to justify realistic amortization would be unwise. Both Soundstream, whose president, Dr. Tom Stockham, developed the first practical audio digital recording system, and DRS, provide digital recording, editing, and mastering for those records where the material seems to justify the additional expense.

Terry then suggested that they demonstrate the results of various sessions and answer technical questions afterwards. I'll comment on my perceptions of the demonstration. Some additional feedback in a later issue of the Speaker from members who were present might be of interest to our non-local readers. (Attempting to describe in print how something sounds is like trying to explain the internal combustion engine to a goldfish. Or like a goldfish trying to explain the internal combustion engine. -- SK)

The first selection was a portion of a Mozart Divertimento recorded in Symphony Hall, Boston, with members of the BSO. The orchestra comprised a wind ensemble plus a (string) bass. Max Wilcox was producer and the record will be available soon on Nautilus. Schoeps omni microphones were used with Trans-amp preamplifiers. Recorded sound was very clear although many in the room had difficulty in hearing either the left or right channel depending upon where they sat. As this was the first selection, most people were desperately trying to adjust their reference point to the questionable music-listening acoustics of the GTE conference room. I suspect this tape sounded very similar, in timbre and in the amount of ambience, to the digital tape from which the Cleveland Symphonic Winds Telarc disc was cut, but with the reverberant character of Symphony Hall substituted for Severance Hall.

The second selection featured the London Symphony Orchestra, recorded in Kingsway Hall, London, where many of their recordings are made. It was a sample from an album to be entitled "Scenes from Sci-Fi Movies" conducted by Etore Stratta for Columbia Records. Two of DRS's Levinsonized B&K microphones were used for the overall pickup and two Neumann U-87's and a U-47 were used to highlight brass and percussion. The selection included a digital edit which was unnoticeable as an edit (no clicks) but some heard a difference in the amount of bass between the two takes (different time and probably different temperature and humidity). The tympani were very clear (low harmonic distortion), and the rest of the orchestra very detailed, but the brass were harsh (perhaps over-miked with a U-87).

The third selection was the same orchestra, in the same hall, playing a section of Beethoven's 9th Symphony. This was a 20-microphone feed done in the usual commercial fashion, and it sounded awful. It was a typical multimike mess where all is shrill and nothing has definition, just like the usual angelcolumbiarca record but without surface noise or breakup on the fortes.

Selection four was a cut from "Bits of Percussion and Jazz," with Farrell Morris; a disc was produced by Audio Directions, 1035 Draughon Avenue, Nashville, Tennessee, and is no longer available. Some first rate sidemen including Stan Getz, Ron Carter, and Kenny Malone accompany a substantial collection of percussion instruments. Sound was very clean, detailed, and somewhat more perfect than the record (of course). Comparing the very well-cut record to other records of similar material, this seems as good as the best. Responding to a question, Terry Tobias commented that one wouldn't expect to hear any artifact caused by the digital process on an analog record because so much degradation occurs in even the best record processing. It would have been interesting to hear that same cut from the record, at this point in the demonstration, but we were to hear a similar comparison later.

Selection five was another recording of the London Symphony Orchestra, conducted by Harold Farberman for Vox. We heard a sample of the last movement of Mahler's 4th Symphony. This was miked with only two B&K microphones and the sound, while clear, was distant and somewhat lacking in impact. Selection six was the same material but with a multi-mike mix somewhat more successful than the previous Beethoven's 9th. Apparently there were fewer microphones and a Studer board was used. Orchestral sound was good, but the solo voices had the usual close-miked sound common in many opera recordings. There is nothing more unnatural than having the orchestra up on stage where it belongs and having the soloist bellowing at you, apparently perched on the shoulders of the person two rows in front of you.

Selection seven was the end of the last movement of Guilmant's 4th organ sonata played on the organ at Church of the Advent, Boston, by Brian Jones. It was loud. Beyond that, I cannot relate to what I heard; there seemed to be more distortion than original information. Having heard what it sounded like in the church and having heard a tape on other monitors where the effect was similar to what I heard in the church, I can only conclude that something was going terribly wrong with the amplified, equalized speakers and the GTE conference room. Perhaps something did not like steady 32' (16 Hz) pedal, mixed with a considerable amount of 16' tone and the trumpet chamade all at once during the last several measures. The result at GTE was like placing your ear two feet from a Skil saw while ripping two-inch oak with a dull blade. Probably not quite the effect Guilmant intended.

The final selection was from Verdi's "Otello," recorded live at the Met. This tape was taken from the broadcast feed before any compression or limiting. The vocal pickup is done with several distant Sennheiser shotgun microphones, pre-aimed and faded up as required by singer's location. Most in the room felt this to be the most impressive sound of the demonstration. This feeling was certainly reinforced by the video: accompanying the sound was a video cassette synched with the audio and displayed on the large Advent Videobeam system installed in the conference room. The dynamic range and sense of presence in the hall (even with one's eyes closed) suggest that microphone techniques distant enough not to obscure audience sight lines are more satisfactory to record classical music. The let-down of the demonstration was the playing of the recent record on RCA of the same passage (end of Act 2). By comparison the sound was thin, lacking in dynamic range, clarity, and bass quality. In fact, it sounded a lot like the NPR radio feed accompanying last year's TV simulcast, and it provided an indication of the amount of damage created for landline/FM TV compatibility. Yet the version on record is really quite good, as a record, very close in quality to the Verdi Requiem with Price and Baker, also on RCA. Who would ever have thought how much was missing? (But on the other hand who has heard an orchestral record that sounds at all like its 15 ips master tape?) Perhaps this was the most significant part of the demonstration: that the ear is easily fooled without a direct comparison. After an hour of digital-quality preconditioning, the jolt back to the reality (unreality?) of analog phonograph records was of a severity similar to coming home from a live concert and putting on a disc of the last part of the program.

Perhaps in response to the dismal sound of the record, there was a clamor from the floor to remove the equalization. The equalization, provided by the Sontec mentioned earlier, was arrived at by obtaining equal third octave response using an Ivie analyzer at the first row of seats, balancing each row of four speakers (each channel) separately. This was done with 100 plus BAS members milling around before the meeting and talking in this area, creating both a high noise floor and continually varying absorptive coefficient. (In addition, see the May issue of the Speaker for why the technique might not work.) At any rate, the consensus from the floor was that even the record sounded better without equalization. Amount of EQ used spanned 16 dB in one channel with a boost of 8 dB between 2 and 3 kHz. Also, a 6 dB level difference measured between the two channels did not sound that way in the room and was reduced during the first demonstrated selection.

Several questions came from the floor, the first being the expected, "Is there any difference between the digitally recorded sound and the original?" Terry answered this by saying that one must compare the digital output only to the input to the recorder; what differences there are appear to vary with the changing of analog portions of the circuitry, i.e., amplifiers, within the A/D and D/A interfaces and do not relate to the purely digital portion of the system. In other words the differences are extremely small and greater differences (degradation) would generally be found in an analog recording system. The question arose of whether the anti-aliasing filters required in PCM systems cause audible coloration. Terry and Peter Jensen feel that the filter used in the PCM-1600 is not a factor, although the situation may differ some with other systems. As for editing ease, Peter felt that editing with the PCM-1600 system was easier than with analog equipment. There is no problem with mechanical integrity, and the ability to preview and continuously adjust each "splice" in a memory before making the final edit allows a better product. As well, a correct final master can be made with no degradation whatever and all the original material remains intact. A question about the difficulty of fitting digitally recorded material onto analog discs, and whether a dbx encoded disc might be helpful for this, elicited the opinion that the dbx disc might be a satisfactory interim measure but eventually digital discs will solve the problem without introducing any side effects.

Terry commented that digital recording is improving standards of recorded sound because of its present limitation to only two channels. Many of the first digital recordings utilized relatively simple microphone techniques, both because there was no opportunity to mix down and equalize later, and because the personnel working the sessions were scientists or design engineers and had some acceptance of the laws of physics. Musicians, audiophiles, and the general public were often enthusiastic about the results, which were initially attributed to the digital recording medium. Then when larger companies hired a digital recording crew to tape their traditional multi-microphone feed, all too often the results were miserable. The message of course is "garbage in, garbage out." (Perhaps with continued reinforcement by the audiophile community and some corporate head-scratching, the message will get through and more traditional methods of sound recording will be applied, at least to classical material. -- SK) The microphone feed is the principal determining factor of the quality of the commercial product, and while digital recording offers some audible superiority for some material, the consumer is unlikely to hear it until a digital disc format becomes available. Much of the superiority the consumer hears with present direct-cut and digitally recorded analog discs is the extra care taken in the analog part of processing to avoid totally ruining what sounds good originally.

What about the difficulty of getting digital material cut onto analog discs? Peter answered this by saying that it seemed to have more to do with the specific material than any problem arising from the accuracy of the digital recording process. In recent years, the cutting process has not limited disc quality. The real limits are found in the playback capability of existing phono cartridges, and in the degradation of groove shape between the original lacquer, via three steps of plating process, and the rapidly pressed vinyl record. (I have not yet heard a final pressing which had the clarity of its reference acetate. -- SK)

Following a brief intermission, John Sullivan of BTX, Inc. explained something about their product line. The controller used during the audio/video part of the DRS presentation is part of a system which provides an SMPTE time code for interfacing audio and visual media. Their system utilizes a 2400 Hz tone giving a synchronization ability of 1/30 of a second. This is equivalent to an 80 bit word per frame. The unit may be used for both videotape and film work and can program edits. When preprogrammed, it will insert edits from three slaves to one master, for example. The controller and synchronizer cost $8500 and the SMPTE code generator is an additional $3500, not practical for home use at these prices.

A local BAS member briefly demonstrated a new turntable by Technics, the SL-10. This is a totally enclosed clam-shell design with the arm/cartridge mounted in the lid and positioned automatically when the lid is closed on the record. The record is clamped down in the center, and the straight line tracking arm is longer than some at five inches -- both factors are a help in playing a warped record. Anew cartridge (Bellex) was demoed briefly with a few records.

The remaining part of the meeting proceeded with a description of the Sony PCM-1 system by Brad Meyer and a further demonstration of several selections of music recorded with this system. The PCM-1 is the poor man's PCM-1600; what with the benefits of modern inflation, one may be labeled "poor" while owning a tape recorder costing slightly less than five figures. The PCM-1 is reviewed elsewhere in this issue.

A footnote: Several aspects of the monitor system provided for this meeting were somewhat unusual -- even questionable. First, for the DRS presentation, the four monitor speakers per channel were placed in rows on two intersecting walls near the floor. The fronts of the cabinets were thus at almost 90^{o} to those of the other channel. The audience chairs were also angled in rows at perhaps 30^{o} relative to the left channel. Then there was the matter of the Sontec equalizer (an extremely fine equalizer, as I have been discovering in the past week; maybe I'll stop soldering together passive networks and use a piece of legitimate hardware for a change). The results of the "room equalization" should perhaps be called "room exacerbation;" when it was removed at the end of the DRS demo, everyone thought the sound was very different and most thought it improved. Actually, the equalizer was not removed from the circuit; but put in "out" mode. Thus the cables connecting it through the system were still in place. The unit uses Cannon connectors (with Pin 3 as hot, unfortunately) and one of the cables used (not Sontec's) was later found to have an inversion. Although the phase problem was detected during the set-up and corrected at the power amp outputs, the incorrect cable wiring may have caused distortion at high levels at the output of the Sontec.

During the intermission, the speakers were stacked one atop the other providing a better point source than the previous arrangement, and the equalizer was bypassed, but still inverting one channel's signal. A brief replaying of the first Mozart wind ensemble selection at the end of the meeting seemed smoother and less shrill than before.

Despite the problems of the GTE conference room for music listening, attending BAS members have learned to allow for it after hearing several different systems over the years. Other listening has always been done with the speakers parallel to a single wall. The intersecting wall placement certainly confused our frame of reference. Yet most had no difficulty determining what sounded good and what did not. Perhaps there is a message here, also. Even a moderately experienced listener, familiar with the sound of live music, can hear good sound versus bad on an unfamiliar system, or on a modest quality system, or on a good stereo car radio. Maybe even on a system without all gold connectors or a Class A power amp. For this reason, those recording music should try to capture something which sounds realistic, because those on the listening end will hear and appreciate it. -- Scott Kent

A Publication of the BAS

The Sony PCM-1: A User's Report

Brad Meyer

Considering how much fuss is being made over digital audio, it's amazing how few people have actually heard a digital recording. While many audiophiles have bought records issued by Telarc, Sound 80, RCA, London, etc. that say "digital" on the cover, those are not digital recordings. They are analog records that have been cut from a digital master tape. They have ticks, pops, and surface noise. They are almost certainly warped to some degree. They must be played with cartridges that can generate distortion amounting to several percent or more in the loudest passages in the inner grooves. They get dirty. They wear out. Phooey on them.

But, you may protest, some of these are the best-sounding records around. Why are you being such a curmudgeon? The reason is that, having heard the real thing, I am much more aware of the degradation's imposed by the disc cutting and pressing processes. A digital master tape, or for that matter a digital copy of a digital master, sounds like the signal that came out of the microphone mixer during the recording session. As you can see from the meeting report in this issue, that does not guarantee excellent, or even good, sound. But it does guarantee you the same electrical signal that the recording engineer was listening to when he decided that the result was satisfactory.

We have all been assured that in a few years we will be listening to digitally encoded material at home. What can we do in the meantime? Several years ago, Sony announced that it would be marketing a product that would take two channels of audio, change them to digital form, and store them on a video cassette. The idea was an interesting one, and the projected price of $1,200, while stiff, seemed within the reach of the serious audiophile. The product was eventually released as the Sony PCM-1 Pulse Code Modulation Adaptor, at the distinctly higher price of $4,400. This figure does not include the video cassette recorder with which it must be used, either. To make matters worse, the PCM-1 is already obsolete. The 13-bit code that it uses has been superseded by a 14-bit system that was recently adaopted by the Japanese audio industry as the "consumer" digital standard. Sony has now made available two machines with 14-bit encoding, the PCM-10 ($5,500) and the PCM-100 (about $14,000).

I have had the good fortune to be able to examine, test, and use a PCM-1, thanks to the generosity of Bob Berkovitz and Acoustic Research. Although nominally "obsolete," the PCM-1 is worth a close look, both because it presents a picture of the general strengths and weaknesses of digital audio and because it is a fine piece of equipment, well thought out, very well built, and fascinating and rewarding to use. I am sorely tempted by the latest version, even at its high price. To help explain why this is, I will begin by giving you some of my own history with tape recorders, so that the context in which the digital machines are operating will be clear.

The Problem

I still remember the first tape recorder I ever saw; it was a Webcor, a bulky, brownish box with rounded corners, very modern-looking for 1954. It recorded and played back in half-track mono at 3 3/4 and 7 1/2 ips. At the higher speed it would reproduce voice fairly well and music

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less well; at the lower speed voices were only fairly good and music was terrible. From this I began to understand a fundamental principle: the more tape you use, the better the sound.

Several years later I got my first good tape machine, an Ampex A-122. The first in a series of technically interesting recorders developed for home use by the well-known pro audio manufacturer, the A-122 was a three-head machine which recorded in half-track mono and played back half-track (and later, when modified with a movable playback head, quarter-track) stereo. The stereo playback feature was included because of the introduction of pre -recorded stereo reel-to-reel tapes, the earliest practical form in which two-channel sound was made commercially available. A close friend had a Capps condenser microphone, and with the Capps and the Ampex we made some concert recordings which still sound good today. On the basis of these early successes, I decided that all one needed to make decent recordings was a good machine and a good microphone. It was at about that time that things started to go wrong; a little learning, etc. .. .

I had agreed to record a Cambridge folk concert in the spring of 1961 (for those who remember, it was Rolf Cahn and Eric von Schmidt). I planned to use a monstrous old Berlant Concertone that ran at 7 1/2 and 15 ips and took 10-inch reels. I figured that because the machine was big, and looked professional, it would do a good job. I recorded at 7 1/2 ips, having reasoned that if my Ampex worked well at that speed, then the larger machine should do even better. In truth, the Berlant was designed to run at 15 ips; at 7 1/2 it was not particularly good. I also went out and bought some tape labeled "high output," naturally thinking it would work better than the stuff I had been using. It didn't. The tape and the machine were severely mismatched, and the highs were so severely rolled off that the recording sounds as though the microphones were wrapped with three layers of wool socks. This spectacular failure was the beginning of my higher education in the realities of tape recording, and ever since then, my knowledge of tape recorders has steadily increased and my opinion of them has been going steadily downward.

It isn't that tape recorders aren't useful. My tape collection contains some great moments, both of music and of recorded sound. It's just that in the process of getting these moments, the tape machine itself has cost me so much time, effort, and irritation that it has come to seem more a necessary evil than a valuable tool. (I have often wondered why I seem to have so many of an object I dislike. Maybe it's because I keep trying to find one that works perfectly.) Of course, if what you want to do is tape records, there are many machines that will do a decent job. But when the material is live music, it quickly becomes apparent that all analog recorders share, to a greater or lesser degree, the following list of faults:

1. <u>They don't have flat frequency response.</u> Magnetic tape is an inherently non-flat medium. Designing heads and electronics that give response that is flat within a couple of tenths of a dB all the way from 25 Hz to 20 kHz would be a very difficult job even if the tape were perfectly consistent, and tape is definitely not consistent, varying significantly sometimes from one reel to another in the same batch, or from one channel to another in the same reel. A conscientious engineer will reset the bias and equalization on his machine for the same few reels of tape he or she will use for a job, and while this gives audibly better results it takes a lot of time to do it right. The problem is of course compounded if you have not two channels to adjust but twenty-four, or thirty-two, or forty. In the real world of professional recording it simply cannot be done as often as it should. Keeping the machine/tape combination operating optimally is a constant battle.

Quite aside from the problem of adjustment there is the question of head shape. Hyperbolic heads of the type used on many home and most professional machines have fewer problems with scrape flutter than heads of circular cross-section, but they tend to cause ripples in the response below 100 Hz or so that can measure as much as 4 dB from peak to trough, and they also roll off the low bass. (This phenomenon is at least partly responsible, in my opinion, for the perceived increase in clarity which some people hear when changing from 15 to 30 ips. The low-end rolloff that begins at 25 to 30 Hz at the lower speed is an octave higher at 30 ips, well up into the range where it will affect a lot of musical material. The attenuation of the very bottom end produces an apparent increase in midrange detail apart from any other benefits of going to the higher speed -- see <u>High-Definition Midrange vs. Deep Bass</u> by R. A. Greiner in the <u>Speaker,</u> July 1978.)

2. <u>Their speed is inconstant.</u> Virtually every tape recorder ever made has audible flutter, if you know how to listen for it. Record a 400 Hz sine wave on the tape, play it back through one

speaker, close one ear, and move your head around until you find a null in the room. Almost all rooms of reasonable size will have enough sharp nulls at that frequency that you won't have to look very hard. As the tape speed varies, the wavelength of the tone changes, and so does the location of the null. This has the effect of moving the null back and forth by your ear, and the resultant changes in level are far easier to hear than the accompanying changes in pitch. This test is probably about five times as sensitive as any test using music, but it gives a useful method of comparison for all machines, including the very best. Solo piano or solo acoustic guitar are not quite as revealing of flutter as the sine wave test, but they are far more so than orchestral or vocal music. There are tape recorders in the $3,500-and-up class that are audibly almost perfect for piano music, but the semi-pro units that sell for under $2,000 have noticeable problems.

There are other kinds of speed variations, too. What is commonly called "flutter" consists of speed variations in the range of 1 to 20 Hz, the ear being most sensitive at about 4 Hz. There is a higher-frequency speed variation caused by friction between the tape and the heads or tape guides, called scrape flutter, which is perceived not as a variation in pitch but as a veiling or lack of clarity of the sound. Scrape flutter affects all types of music, especially orchestral and choral. Finally, there are long-term variations in speed that affect most recorders, especially those that take ten-inch reels. Anyone who has tried to splice together two sections of music from widely separate places on a reel has discovered that the tape slows down slightly as the reel progresses, because the take-up tension decreases while the hold-back tension increases, as the two reels empty and fill. Only the most expensive professional machines with tape tension controls are free of this problem. Neither is there any guarantee that speed will be audibly identical from one machine to the next; I have observed speed differences of almost one percent between semi-professional machines each costing over $1,000. On machines with electronic speed control, which is generally a desirable feature, the speed can change as the machine warms up and the components in the speed control circuit change their values.

3. Their system gain varies with time. In its most well-known form, this problem is known as drop-out. But smaller variations in playback level occur all the time, especially in the high frequencies. Add a noise-reduction system that uses some form of companding, as all of them do, at least above 4 kHz, and things get even worse. Even though most of this variation in level occurs over a range of one dB or less, I believe it contributes to a feeling of unsteadiness, or lack of solidity, which helps identify the sound in the mind of the listener as artificial. Like flutter, this problem is most audible on steady tones, and almost as much so on piano music.

4. They are too noisy. There is a lot of talk going around about how digital recording will at last give us "the full 90 dB dynamic range of an orchestra." This seems a good time to clear up that misconception. If you take a sound level meter with you to the next concert you attend and look at it during quiet passages, you will discover that at no time when anyone is actually playing will the meter read less than about 50 dB SPL, and that the average soft passage will be in the high 60s or low 70s. Measurements I have made with a peak-reading meter, using a microphone above the stage and closer to the orchestra than the first row of seats, have not yet produced any level above 115 dB; taking that as a maximum, the dynamic range of a 100-piece symphony orchestra playing a work that encompasses the loudest and softest that the players can manage, like the Mahler Ninth Symphony, is about 65 dB. A good conventional tape recorder with modern high-output tape can record levels of 10 dB or more above the 0 VU level of 200 nano-Webers/meter, and will have a noise floor about 65 dBC (C-weighting approximates a 20-20 kHz passband) or about 72 dBA (rolled off below 500 Hz and above 4 kHz, more or less) below the maximum output level. So an analog recorder can record the full dynamic range of a symphony orchestra.

The important distinction, and the source of the confusion, lies between dynamic range and signal-to-noise ratio. When the violins are fading slowly out at the end of the Mahler Ninth, and the enraptured audience sits absolutely still, scarcely breathing, the noise in the hall, if it is properly designed and built, may be as low as 25 dBA SPL. The equivalent noise of the tape recorder, meanwhile, is about 43 dBA. The s/n of the music on the tape at that point is less than 10 dB, and the acoustical background level is nearly 20 dB below that of the tape. This is the reason that noise reduction systems were invented.

Though noise-reduction systems all help the problem for which they were designed, each has its drawbacks. The Dolby system depends on the correct calibration of levels in the record/

playback chain, and creates audible errors when it is miscalibrated. This happens in cassette recorders when different types of tape are used without recalibration, and it happens a dismaying amount of the time in professional noise-reduced tape recordings through engineering carelessness. The most popular alternative, dbx, works well for most types of music, but can give trouble on piano, harpsichord, or guitar because after removing the tape hiss there is still modulation noise, which rises and falls with the level of the music and cannot be separated from it. dbx can also produce audible noise-pumping during soft passages in certain kinds of wide-range material.

5. <u>Their distortion increases with recording level.</u> Actually, the distortion in analog tape recorders is mostly third harmonic, which is not especially harsh-sounding, and when these machines are driven past the point of tape saturation they overload in a way that is advantageous in some circumstances. But tape recorders have IM distortion too, and on massed brass or choral music this can cause muddying of the sound if record levels are not kept conservatively low.

<u>The Solution</u>

It should not be necessary to give more than a brief description of the digital recording process at this point. The essence of the concept is that the musical signal is converted to a string of binary numbers, and the <u>numbers</u> are stored on the tape. During playback, the numbers are read off the tape and converted back into a facsimile of the original signal. To some audiophiles, the use of this facsimile means that the signal <u>must</u> be audibly degraded; whether or not this is true, the processes of conversion into digital, and from digital back to analog, are the crucial steps, and how well they are performed will determine the quality of the final result. This is so because storing (and retrieving) binary numbers on magnetic tape sidesteps almost completely the problems we discussed in the previous section. All the tape recorder has to do at any given instant is to deal with one of two levels of magnetization, one representing 1 and the other 0. If the system can do this without error, the number can be stored and retrieved with perfect accuracy. The problems of distortion, noise, poor frequency response, and small level variations common to analog tape recorders are almost completely irrelevant to this task. The other problem, inconstant speed, is taken care of by reading the numbers off the tape into the d/a converter under the control of a quartz crystal clock. The same clock is used to encode and decode the signal, so the speed accuracy of the system is equal to the accuracy of the clock. Translated into conventional specifications yields a flutter figure of something like 0.003%, which is inaudible under any circumstances, with any signal.

Aside from the possible difficulties of the encode/decode operations, the big disadvantage of digital encoding is the demand it places on the high-frequency performance of the tape. The accuracy of the digital approximation is a function both of the rate at which the signal is sampled and quantized, and the length of the binary word used to represent its value. Digital audio requires a bandwidth of from 1.5 to 4 MHz, higher by a factor of 100 or more than analog machines. Fortunately, there already exists a family of tape recorders that has been designed for extended bandwidth at the expense of other parameters: the half-inch and three-quarter-inch video cassette recorders that are becoming more and more common in both home and industry. If you have a VCR, you can use a digital adapter.

<u>The Device</u>

The PCM-1 is unusually large and heavy for a piece of all-electronic gear. Its dimensions are approximately 19" x 16" x 10" (w x d x h) and it weighs over 40 pounds. The chassis is made of heavy-gauge sheet metal, and can easily support a VCR on top. The front panel is dominated by two large, heavy knobs which control the record level for the two channels. Near these is a set of gas-discharge peak-reading level meters with a range of -42 to 0 dB, with an additional pair of lights indicating overload. The meters are covered with an impressively thick piece of glass with beveled edges. A smaller knob near the meters selects "peak" or "peak hold" operation; the "peak hold" mode retains the highest reading during the time it is engaged, up to but unfortunately not including the overload lights. There are two phone-plug (unbalanced) microphone inputs near the lower right corner of the front panel, and above them a selector switch to choose between line inputs, mike inputs, or mike inputs with an attenuator. There is a two-position switch labeled "pre-emphasis" which supplies a high-frequency boost during recording; a signal is recorded on the tape that tells the playback section whether the pre-emphasis has been used, so the switch is inoperative during playback. A power switch, a headphone jack, and a headphone level control complete the front-panel list.

On the back, the PCM-1 has a pair of line-level inputs, and a corresponding pair of outputs. There are two sets of these, one (unbalanced) using pin jacks, and one (balanced) with cannon connectors. Inexplicably, the cannon connectors are the wrong sex: the inputs are male and the outputs female. (This error and the failure of the peak-hold mode to include the overload lights are the only design mistakes I could find.) Finally, there is a pair of pin jacks, one labeled "video out," the other "video in." To hook up the adapter, connect the video output of the PCM-1 to the video input of the VCR and vice versa, make your line (or mike) input and output connections for the audio signals, switch the VCR to "camera" (or "aux"), and you're ready to go. The level meters and the monitor circuit in the PCM-1 use the video return from the VCR as a way of making sure that everything is working properly. As a result, on some machines (e. g., JVC) you must depress the record button (but not, as one reviewer has claimed, actually run the tape) in order to set levels or get an audio output.

I didn't measure the noise level of the PCM-l's mike preamps, but they add audibly to the noise if turned more than half-way up, so the attenuator should be used only if the incoming signal is unusually high. You set the meters on "peak hold" and get the musicians to play loud, and set the levels to read a maximum of about -3 dB. Using peak-hold lights is a different experience from watching ordinary mechanical needles. This is especially true for piano, which has very loud, brief attack transients even when it is being played softly. If you see -10 dB peaks during a mezzo-forte, you won't overload the machine even on the most thunderous climaxes. But it is necessary to be careful when getting levels to make sure that the system <u>never</u> overloads. Unlike analog tape, which clips softly and almost inaudibly when mildly overdriven, a PCM system clips abruptly when it reaches its maximum voltage. This can make ugly noises if it is allowed to happen more than very briefly, so a conservative hand on the level controls is necessary. The professional PCM-1600 has an arbitrarily defined 0 VU point some 16 dB below its maximum level, so that professional users can treat the zero the way they are used to doing on their analog machines without running into trouble.

<u>On the Test Bench -- Digital versus Analog</u>

In the following series of tests we will be looking at three recording systems. To help place the PCM-1's performance in the context of existing tape machines we'll compare it to a half-track, 15 ips semiprofessional tape recorder with unusually flat response, the Revox A-77 Mark II. The third "system" consists of the same Revox with the addition of a dbx II noise reduction unit. The latter setup is the one I have been using for most of my mastering for the past three years or so.

Figure 1 shows the record/play frequency response of the Revox as adjusted for the particular reel of Ampex 456 Grand Master used for the tests. The adjustment was done in about five minutes; no extraordinary effort was needed to achieve this result. The vertical scale of this chart gives it almost exactly the same proportions as the frequency response graphs used in High <u>Fidelity</u> and <u>Stereo Review.</u> As you will see, changing the vertical scale can change almost completely the way such a graph looks. These proportions make even a mediocre cassette deck look okay, and a really flat machine like the Revox looks superb. The A-series Revox is in fact unusually flat even compared to other professional machines, in part because its playback head, which as seen from above is shaped like a segment of a circle, has fewer and less severe aberrations at low frequencies than the more common hyperbolic heads. At 7 1/2 ips the Revox is flatter in the bass and slightly rolled off in the top octave.

Figure 2 is the Revox/dbx combination. The slight response errors visible in Figure 1 have been approximately doubled by the compression/expansion process. It should be pointed out that the system does not have this degree of error on musical material; for instance, if the dominant tone in the music which determines the response of the dbx circuitry is at 220 Hz, where there is a slight rise, then the error will show up as a level error of +2 dB across the whole spectrum, not as a peak at 220 Hz.

Figure 3 is the PCM-1's frequency response. It appears literally ruler-flat from 20 Hz to 5 kHz, above which there is a very slight droop which steepens a bit at 15 kHz before coming up again, then sharply down above about 21 kHz. The shape of the curve at the very top end reveals,

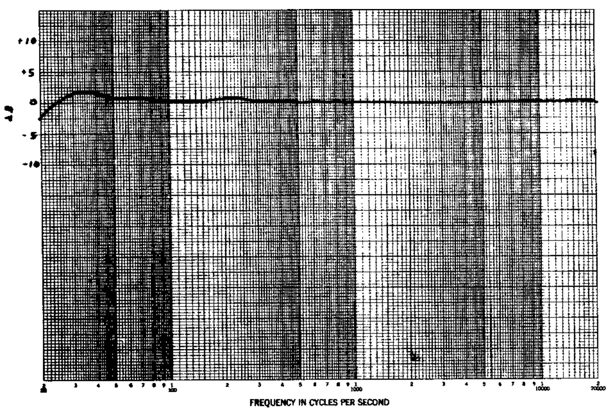

Fig. 1. Frequency response-half-track, 15 ips tape

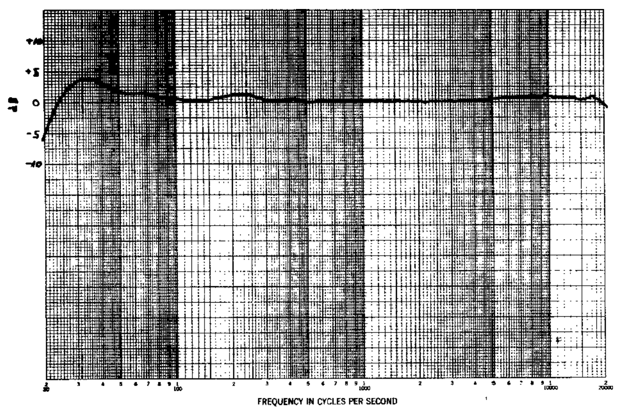

Fig. 2. Frequency response-half-track, 15 ips tape with dbx II noise reduction

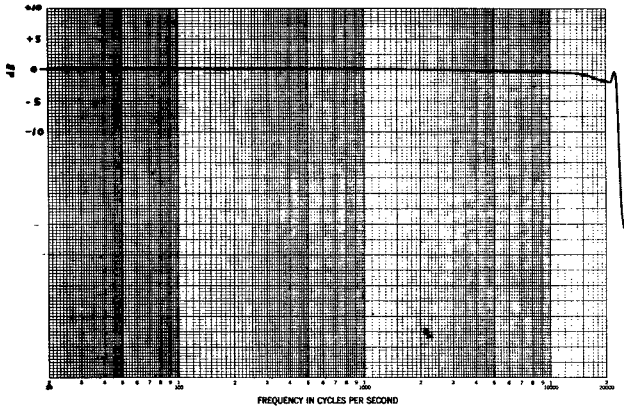

Fig. 3. Frequency response-Sony PCM-1 (pre-emphasis off)

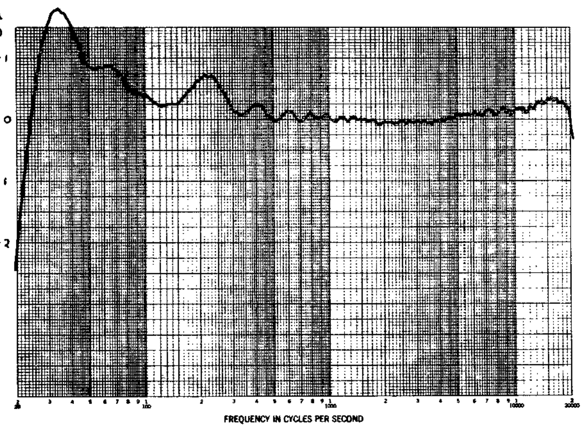

Fig. 4. Half-track, 15 ips tape (expanded vertical scale)

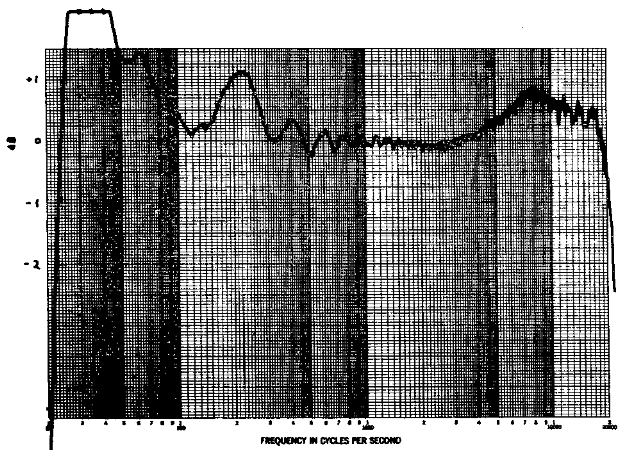

Fig. 5. Half-track, 15 ips tape with dbx II (expanded vertical scale)

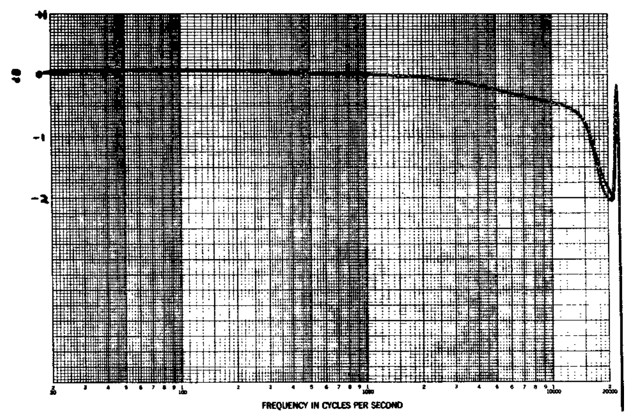

Fig. 6. Sony PCM-1, left and right channels superimposed (expanded vertical scale)

at least to the electronic designers I've shown it to, that the anti-aliasing filter is an elliptical design. As you can see, the rolloff is very steep. A low-pass filter this steep is necessary in a pulse-code modulation system because if there is any material above the Nyquist frequency, which is half the sampling frequency, the decoder can't tell it from a signal at a corresponding distance below the Nyquist frequency. In other words, for a sampling rate of 44 kHz, and a Nyquist frequency of 22 kHz, ultrasonic garbage at 32 kHz would come out of the decoder as 12 kHz. This obviously won't do, so eight-pole or higher filters are used. These filters inevitably have some phase shift in the audible band, giving those who don't like digital, and who hold that the phase shift will be audible, some support for their arguments.

Figure 4 is the Revox again, with the vertical scale expanded by a factor of ten. The minor head bumps at the bottom end suddenly look major, and can be seen to continue up into the midrange. There is also a slight rise above 5 kHz. What appear to be ripples in the top two octaves are actually a sign of less than perfect tape-to-head contact; you would see them if you plotted the output of a steady 10 kHz tone.

Figure 5 is the expanded-scale plot of the Revox/dbx combination. As before, the frequency response errors are artificially exaggerated, but so are level errors resulting from tape to head contact, and the latter may actually make an audible difference. Tape quality is definitely more critical with dbx, for this reason.

Figure 6 shows the response of the two channels of the PCM- 1 with the same scale as 4 and 5. This plot was made with the high-frequency pre-emphasis off; for some reason the system is flatter with it on, the way it is actually used. With pre-emphasis the device is down only 0.25 dB at 10 kHz and 0.5 dB at 15 kHz. The behavior of the filter is clearly seen, and what is most remarkable is the degree of match between the two channels. The filters in the PCM-1 are little gray modules, encapsulated in epoxy. There must be someone at the Sony factory who sifts through a bin of them to find a pair that match for the two channels of each machine. This may be important enough to justify the trouble of doing it, because whether or not you believe that the absolute phase shift of these filters can be heard, a relative phase shift between channels can affect the stereo image. Rumor has it that not all PCM-1s roll off above 15 kHz; some supposedly have a slight rise, and some are flat. But it looks as though inter-channel phase is maintained to close tolerances, something which cannot be said of analog machines because of mechanical problems such as tape skew.

The next three figures show several things. One is the "signal space" afforded by each system. By this I mean the area between the overload limit of the system on top and its noise floor on the bottom, as a function of frequency. Also displayed is a one-third-octave analysis of each system in the presence of a 1 kHz tone. This reveals the extraneous distortion and noise generated by the system in the presence of a signal. The test tone is at 0 VU on the Revox's record meter, which because the Ampex is a "hot" tape turns out to be 276 nanowebers/meter, 2.8 dB higher than the industry standard 0 VU (and open-reel Dolby) level.. The PCM-1 was tested at -10 dB re maximum level at 1 kHz, and the vertical axis of the chart is marked with -10 dB as the reference. This allows approximately 10 dB of headroom in each system above a nominal maximum steady-state level, to allow for brief musical peaks. Although -10 dB is the closest even value to the correct level, the peak factor for actual music is generally greater: if you dub music from an analog tape onto the PCM-1 and set the maximum level at -2 dB with the Sony's peak-reading meters, then play a 0 VU tone on the analog machine, the Sony's meters will read about -14 dB.

At the top of each graph are the maximum outputs of each system versus frequency. For the analog system the maximum is the level of 3% third-harmonic distortion at frequencies up to 4 kHz; above that, it is the level at which the input/output ratio becomes nonlinear. (Cassette recorders are frequently measured for headroom at high frequencies with a twin-tone IM test, which involves putting two tones 1 kHz apart on the tape and increasing their level until the 1 kHz difference frequency reaches 3% of the combined test tones. This is not a proper test for open-reel half track tape at 15 ips; at 16 kHz the tape saturates at +7 dB, but 3% twin-tone IM occurs at an input level of +16 dB.)

You will notice that the maximum levels for the dbx tape are the same as for the Revox alone. Doing the test with steady-state tones would yield a higher overload curve (more headroom) with the dbx, but this would not give a true indication of the capabilities of the system on music.

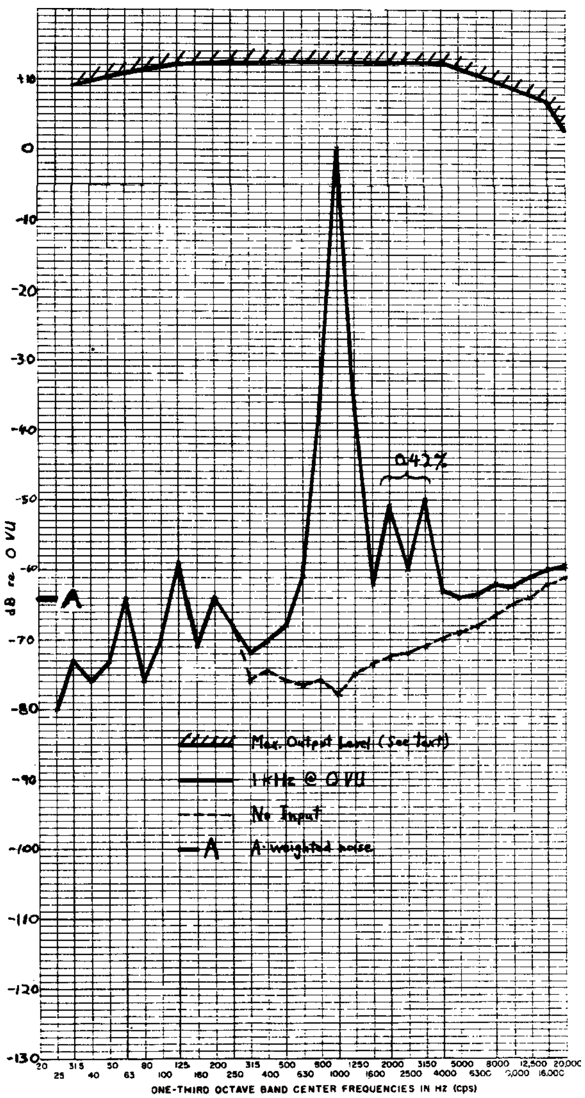

Fig. 7. Half-track, 15 ips tape (Ampex 456) no noise reduction

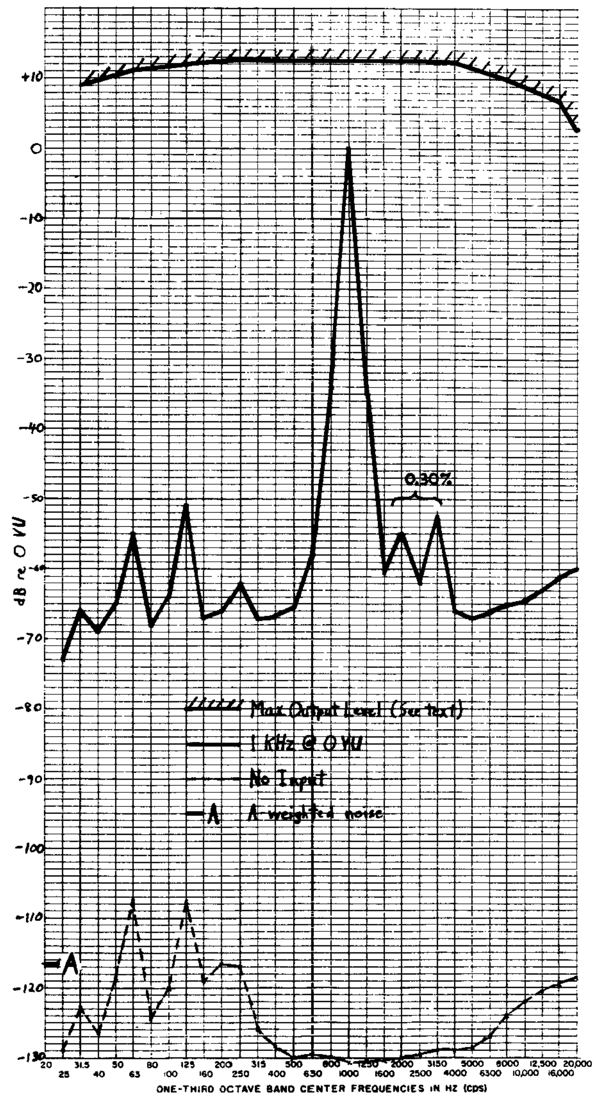

Fig. 8. Half-track, 15 ips tape with dbx II noise reduction

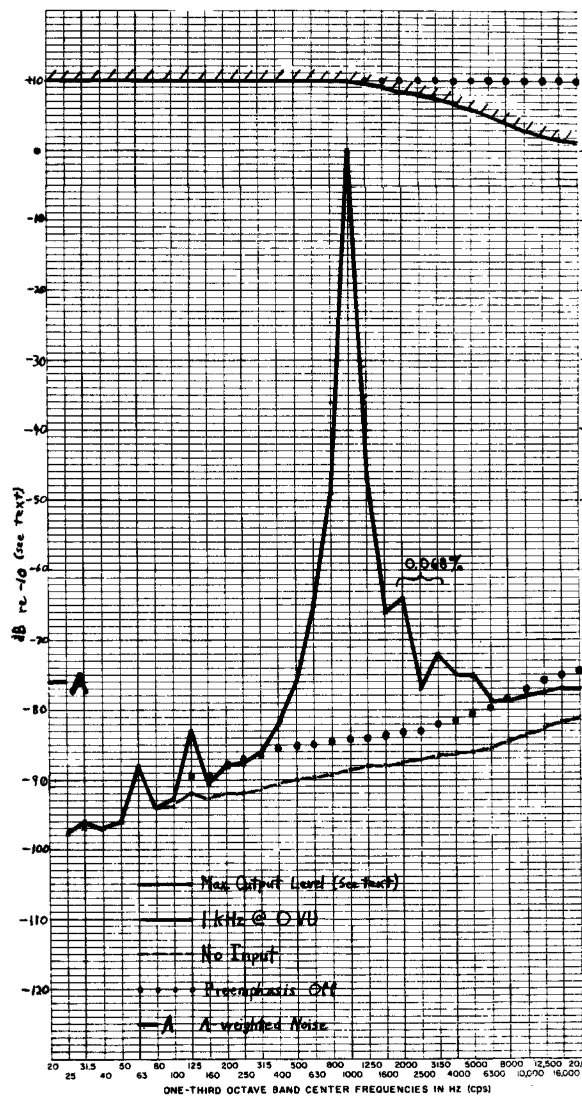

Fig. 9. Sony PCM-1 digital adapter

Putting a peak-reading meter inside the dbx loop while recording piano or choral material reveals that the peak/average ratio of such material is almost unchanged by the dbx, because its detectors are not fast enough to compress very short peaks. Since it is therefore necessary with some kinds of music to keep the Revox's meters at the same place as when using no noise reduction, the effective overload point is shown as being the same.

Figure 7 shows the performance of the Revox alone. A notch filter providing about 35 dB of attenuation at the fundamental frequency was used in measuring the spectrum denoted by the solid line, so the 800 and 1250 Hz bands really do give some indication of the amount of modulation noise around the test tone. At low frequencies the noise is the same with or without signal, and there is a slight increase in hiss with the tone present. Second and third harmonic distortion products are nearly equal, and total 0.42%. Headroom is very generous, which is the strong point of Ampex 456 (or Scotch 250), and is in fact limited by the Revox's record electronics between 250 and 2000 Hz. (This is why people take their A-77s to Scott Kent for modification.) Also shown at the left edge of the chart is the A-weighted noise level for no input. The dynamic range of the system stated, as is standard practice for tape recorders, as the difference between the A-weighted noise and the 3% distortion point at 333 Hz, is 76 dB. This is a very respectable S/N, needless to say.

Figure 8 shows the effect of adding dbx. Companding systems give greater improvement the better the machine is to begin with. The machine is very good to begin with. The noise spectrum with no input shows what happens on a dbx tape between takes -- the hiss disappears, utterly and completely. You have to wrap your ear around your tweeter to hear anything (very uncomfortable if you have horns or electrostatics) and even then you can't be sure it isn't just the high-level section of the preamp. The signal to noise ratio comes out to a mind-boggling 129 dB!

This effect is great for wowing your friends, and it sells a lot of dbx units, I'm sure. But as you can see from the spectrum analysis of the test tone, it doesn't reflect what happens in the presence of signal. At 0 VU the modulation noise and distortion are about the same. There is high-frequency pre-emphasis in the record section of the dbx, and the corresponding playback de-emphasis reduces the distortion products by about 3 dB, while increasing the hum by 6 to 8 dB. Most of the time the music masks this background noise, as it is supposed to, but the tape hiss varies so rapidly with signal level that at occasional moments, such as when a single soprano or clarinet is heard during a large-scale production, the hiss or hum can be heard varying in the background. This may also be partly due to the modulation noise, which, dbx claims, becomes audible when the hiss is removed. The modulation noise problem frequently occurs in recording solo piano in a very quiet environment, too. Interestingly enough, dbx-encoded copies of piano music made from B-Dolby master tapes or direct discs are free of hiss pumping. It seems that with a certain minimum amount of background noise the large, distracting variations in background hiss don't happen and/or the modulation noise is masked.

Figure 9 is the PCM-1. Overload levels and noise floor are shown with the pre-emphasis on, the way the system is virtually always used; the unconnected dots show the noise floor and overload level with the pre-emphasis off.

The overall signal/noise ratio is better than the straight analog tape, at 86 dB. (This noise level turns out to be audible on extremely wide-range material. The extra 12 dB afforded by the professional systems seems to be useful in eliminating the last vestiges of noise.) The noise floor consists not of digital quantizing noise, but of white noise which is injected deliberately to hide it. Digital noise has a nasty ripping sound, and would be more obtrusive by itself even though its level is lower. As you can see, the hum of the system is also extremely low, far lower than analog tape and far lower than the low-frequency noise found on the best discs. Perhaps most significant is the modulation noise and distortion, which are 10 to 20 dB lower than either analog system in the presence of high-level signals.

You will notice that the amount of headroom you lose by using the pre-emphasis is less than the amount of improvement in the noise. This is the tip-off that something trickier than mere treble boost/cut is happening when you turn that switch. There must be additional companding, almost certainly in the digital domain. The PCM- 1 already uses a form of digital companding known as gain-ranging. The system is what is called a 12+1 bit device, meaning that a 12-bit word and one bit of gain information are recorded. This gain-ranging is instantaneous, meaning that any time the signal reaches a peak level of -12 dB, the gain of the system changes by 12 dB. If you put in a 10 Hz tone whose peak level is, say, -10 dB and look at the distortion products, you see a tiny glitch every time the waveform crosses -12. The effect is barely audible even listening to the distortion signal alone, and is completely inaudible under any conditions of actual use. Our test tone, with its peak level of -10 dB, spends 60% of its time above the -12 dB level, and the increase in the noise above the no-signal value above 5 kHz reflects this. There are no audible effects of gain-ranging on any musical material I have heard, including some which instantly reveals the non-instantaneous gain ranging in some other commercial PCM equipment.

Dropouts

All of the results of the previous bench tests hold true only if the tape recording system used with the PCM-1 is functioning properly. Because the tape has the relatively simple task of reading one of two states of magnetization, its job is much easier than the one an analog machine must do. But a dropout in an analog system manifests itself only as a period of reduced output. In a digital system, insufficient output causes the decoder to lose track of the stream of bits, so the audio signal disappears entirely, and when the signal is picked up again the instantaneous voltage may be very different. The transition between the two will cause a click if the dropout is short, and if the dropout is long the system may put out a burst of very high-level noise. For this reason, the PCM-1 has a muting circuit which cuts in if the bit stream is interrupted for longer than a few milliseconds. This circuit operates very rapidly and silently.

Professional digital adapters record the data at more than one place on the tape, so that if a large dropout occurs the missing bits can be retrieved from the alternate location. This technique is known as error correction, and there are many different ways to implement it. The PCM-1 has a simpler system, called error concealment, which mutes the signal in the case of a long dropout, and for a short one tries to make a smooth transition between the "broken ends" of the signal voltage. In practice this produces a click, about like the sound of a small record tick. There seem to be two kinds of dropout: one, the more common of the two, is not produced by a defect in the tape, but by some other playback difficulty such as skewing of the tape, poor tape-to-head contact, etc. The evidence for this is that these dropouts do not happen in the same place on the tape every time. The second kind, which is repeatable and therefore tape related, is quite rare; I have encountered it on one or two out of perhaps forty different tapes I have used. An attempt to look at the problem area by carefully pulling the offending length of tape from the cassette revealed nothing -- to the naked eye the tape surface was perfect.

If the VCR is properly matched to the adapter, audible dropouts seem to occur at the rate of three or four per cassette, on the average. But many cassettes are error-free; with a chart recorder one could presumably check each one from beginning to end before use. There are companies which provide this service for those who need dropout-free video tape. There is frequently a burst of dropouts at the beginning of each new recorded section, so it is necessary to start the tape five seconds or so before the program begins. Tight cues or punching into record while the tape is moving are not possible.

The frequency of inaudible dropouts, though, is actually quite high. The newer Sony PCM-10 adapter has three different kinds of error correction or concealment, and little lights that blink when each one works. I have seen one operating with a recent model Betamax deck, and the light flashed an average of one to two times per second. You can get the same information from the PCM-1 by recording a tone at about -30 dB and listening to the output with headphones at high level. There are faint clicks, inaudible on speakers, which seem to signal when the error concealment circuit operates, and they occur (with my VHS machine) at nearly the same rate as the PCM-10 was showing. Incidentally, it's a good thing digital hardware is not too fussy about frequency response errors and distortion on the tape, because the output waveform of a VHS or Beta recorder is a very poor replica of the input. (Three-quarter-inch machines are much better.)

Compatibility

So far I have tried the PCM-1 with half-inch VHS recorders made by Matsushita (Magnavox, Panasonic) and JVC, and with three-quarter-inch U-matics by JVC and Panasonic. The only severe problem I have had was with a JVC model 3600 VHS machine. I had a recording commitment at the same time that a guest needed to see a videotape at home, so I borrowed the JVC from an acquaintance. I figured that the JVC's allegedly better picture meant that it would work better with the digital adapter than my Magnavox, and besides, the JVC was smaller and lighter. So I sent the JVC out on the job and kept my machine at home. The recording made on the JVC was filled with dropouts, both short and long: three or four clicks per second, and frequent gaps when the muting circuit cut in. In desperation I took the PCM-1, some tapes, and a pair of headphones down to the New England Video Center store near North Station and tried practically every machine in the joint. The JVC 3300 and 3600 in the store would not work with the adapter either. Finally I discovered that a certain JVC portable VHS machine would play back the original tape with almost no dropouts, and that the tape could be copied directly onto a Panasonic portable such that the copy would play back on almost anything in the store. (This copy was the piano tape played at the BAS meeting.)

At first it seemed as though the problem must be flutter. Al Foster has measured two JVC machines and found them to have flutter of over 0.5%; perhaps the flutter was so great that the input buffer in the adapter was occasionally either overflowing or emptying. The portable machine was advertised as having exceptionally good tape motion control, and it seemed possible that its servos might be good enough to follow the control track on the master tape so as to reproduce exactly the original flutter, cancelling it out. (Another VHS machine that Al measured, an RCA made by Matsushita, checked out at around 0.15%.)

The answer came when I rented a Panasonic professional U-matic recorder for a mastering job and found that the original "unplayable" tape could give perfect results if the video output of the Magnavox was routed through the U-matic's electronics before going to the adapter. It seems that the PCM-1 is quite sensitive to video level, particularly at the highest frequencies, and that too <u>much</u> signal will cause trouble as well as too little. One way the early JVCs got their reputation for superior picture quality was by using lots of high frequency compensation to correct for the losses in the record/play process. Visually, this increases the resolution. But the signal was too much for the adapter. The U-matic machine had a video level control and a meter, and it was possible to read the range of levels over which the PCM-1 was happy. (Unfortunately, the meter was not marked with a quantitative scale.) Further investigation revealed that the range of acceptable levels decreases as the high-frequency losses increase; apparently the better the waveform, the better the system works.

You might think that this would rule out the use of the longer-playing speeds on the video recorder. It turns out that the four-hour speed works just as well as the two-hour on my machine, in both sonic performance and freedom from dropouts. The key fact is that the tape-to-head speed in a video recorder changes virtually not at all at the longer-playing setting, because it is determined not by the tape motion past the head assembly but by the rotation of the drum that carries the heads. Lowering the tape speed merely pushes the angled tracks closer together, and apparently the narrower track does not change the dropout rate much.

A brief encounter with Dr. Toshi Doi, Sony's chief digital designer, revealed that the VHS and Beta systems use different methods for controlling video level, and that the PCM-1 cannot be expected to work with VHS machines. That it does so quite well is attributable to the state of adjustment of VHS machines as a class; with Beta decks there is a part of the video waveform which allows the PCM-1 to correct for a small level mismatch. (The successor to the original model, the PCM-10, is designed to work with both Beta and VHS machines.)

<u>Editing</u>

These adapters have one large drawback for any purpose other than concert recording: they have no provision for editing. Digital tape cannot be spliced, and if it could, the splice point would produce a dropout of unacceptable size. All editing on digital machines (with one exception, which will be mentioned later) is done by selectively copying from one machine to another, like normal video editing. Sony's professional units, the PCM-100 and PCM-1600, are designed to do this efficiently, capable as they are of playing one tape while recording another. On the PCM-1, copies must be made directly from one deck to another without decoding, which means that any dropouts are copied instead of being fixed by error concealment. Any "splices" must therefore be made between movements, where the momentary disappearance of room sound will be relatively innocuous.

If the recording is done on a professional U-matic machine, you can use existing video editors which can copy from two machines to a third and make splices from one playback machine to the other with a precision of 1/30 of a second. More often than not, however, this produces a click if done during the music. What you can do is make digital master tapes with the PCM-1 which can be copied later (decoding and re-encoding) onto one of the professional systems; once the raw takes are in the pro format you can make inaudible edits in the digital domain. The Sony professional editing system is terrific: a small, desk-shaped thing with remote controls for three U-matic decks, which stores six seconds of digital signal from each machine and allows you to mark the edit points in the two samples to some absurd degree of precision, then listen to the edit you have chosen before telling the tape decks to execute it. There is a knob about four inches in diameter with a little hollow near the edge to put a fingertip into. Rotating the knob cycles through the six-second sample just as though you were rocking the actual tape back and forth, while the digital time display keeps track of where you are to the nearest 10 milliseconds. One slight drawback to this toy is its price: $50,000, less recorders.

Subjective Test Results -- So How Does It Sound?

How the PCM-1 sounds depends on the type of music you're listening to. On orchestral or choral material, where there is energy over a broad range of frequencies, the distortion and noise of an analog system are pretty well masked by the music, just the way it says in the dbx owner's manual. The ultimate signal/noise ratio of the noise-reduced analog system is such that it is possible to be very conservative with levels, so distortion is kept low. In addition, most orchestras and choruses have enough variation in pitch, both from one musician to another and from instant to instant for the same musician, that the flutter in the analog system is inaudible. So for most types of ensemble music, the PCM-1 sounds about like a very good analog system.

BUT -- when you record a solo piano or guitar, the advantages of digital become apparent. The absence of hiss or modulation noise makes the sound noticeably cleaner, and the complete lack of flutter gives a solidity to the sound that you didn't know was missing, because years of listening to low but subliminally audible flutter have taught you to expect a certain amount of unsteadiness as normal. There is also a cumulative effect of the fact that the frequency response of the digital system is exactly the same every time you use it, regardless of what type of tape is in the machine or how much time you have to set things up. Unless the video recorder suffers a major malfunction, or is mismatched to the adapter, the system gets set up and plugged in, and it works perfectly. What happens after a while is that a certain hardness or minor coloration which you previously ascribed to the recording system becomes clearly the fault of the microphones or of their placement. There is no longer an easy explanation for such problems; they become harder to dismiss, forcing you to confront the weaknesses in your equipment and/or technique. Soon it becomes obvious, in case it wasn't already, that there are no microphones that sound really natural, even if they cost hundreds or thousands of dollars; there are only different sorts of colorations, off-axis response aberrations, etc.

There is one part of the spectrum that the digital system handles far better than any analog machine: the bass. The PCM-1 is absolutely flat down to well below 10 Hz, and has virtually perfect phase response at 20 Hz. While I am not one of those audiophiles who insists that these criteria must be met for normal musical material to sound right, there is a noticeable improvement in some cases in the way the feel of the room is captured, if your system extends below 30 to 35 Hz, on recordings made with pressure-type (omnidirectional) condenser microphones. (This was demonstrated at the BAS meeting with some of the most faithfully recorded traffic noise ever.) There is never a trace of boominess or muddiness from the system, although as I mentioned above, you soon discover that it is possible to generate these problems by improper mike placement, especially in small rooms.

At the Edges of Perception

Subjectively speaking, evaluating the PCM-1 is more like judging a preamp than a tape recorder. The only flaws in the simile are the effects of dropouts and the limitations imposed by noise. A well-designed modern high-level preamp section has a signal-to-noise ratio about like the dbx-encoded tape in Figure 8, but of course without the noise fluctuation with signal level. The PCM-1's A-weighted noise floor is about 40 dB higher than that, so noise can be a problem unless the levels within the digital recording loop are carefully watched.

As we have seen, tape recorders as a class have hiss, or hiss pumping, modulation noise, flutter, scrape flutter, poor tape-to-head contact, etc., whose cumulative effect is what subjective reviewers like to call "veiling." Veiling is a subjective term which means anything that gives the listener the feeling of hearing the music in a less than direct way, or that produces a mental image of a curtain or other filmy object covering the sound. Veiling can come from many different sources, including background noise in the signal channel, distortion, small rapid variations in signal level or pitch, and subtle variations in frequency balance, particularly those producing a slight downward tilt of the frequency response curve with increasing frequency. (It is a current pet theory of mine that large errors in frequency response are perceived in a comparatively straightforward way as colorations, while very small errors tend to masquerade as veiling, edginess, lack of "aliveness," and so on. If this theory turns out to be true, it could explain much of the "subjective versus objective" controversy about the sound of electronic components. Obviously, more work needs to be done.) Of these possibilities, distortion is the most popular explanation, but is in my opinion most often incorrectly singled out as the source. Often overlooked are the effects of background noise, either in the signal channel or in the listening room. The soft whooshing noise of a cooling fan, or a power amp which emits mechanical hum, or traffic noise coming in through windows or walls, can often cause veiling without even impinging on our awareness. This is one reason systems often sound better at night. Perhaps a useful subjective definition of veiling would be "anything that requires the expenditure of mental energy in order to listen through it." By this definition, the PCM-1 is a superb recording system. It doesn't matter what you use to listen to it, including the fanciest electrostatic headphones. The output of the system simply sounds like the signal coming from the source, plus the noise at -87 dBA. And with most material, the level at which you wind up listening makes the noise inaudible.

There is one thing in the test results which has doubtless caught the notice of the sharp-eyed epistemologists in the crowd. The PCM-1 has a slight but definite rolloff of the high frequencies due to its antialiasing filters. My experiments with electronic components have led me to believe that very small differences in frequency response produce audible effects, whatever else may or may not be going on, and I confess to being puzzled as to why the variation in Figure 6 is so hard to hear. Most of the time, it escapes me totally; once or twice, with good speakers (not headphones, which seem for some reason less sensitive to small frequency response errors) and live source material with significant content above 10 kHz (e. g., a small baroque chamber ensemble, playing original instruments with very delicate-sounding upper overtones) I have heard the subtlest kind of dulling of the sound. My overwhelming conclusion has been that with this particular digital system the recording machine is good enough to stop worrying about, because there is so little room for improvement that it's a much better use of everyone's time and energy to improve other things in the chain instead. In the example above of the baroque chamber group, I had been listening to the monitors for several sessions, thinking we were getting a very nice recording, when finally I went upstairs to the performing hall to listen to a live take. I came back downstairs feeling very discouraged, wondering whether there was any way to build a microphone that really works.

Much has been written recently in the audiophile press about the alleged evils of digital sound. A great deal of it comes from people who aren't technically sophisticated, but who are offended by the idea of chopping the signal into discrete bits and then reassembling it, as though it were so much hamburger. (I mean, hamburger doesn't taste like steak, does it?) One visually-oriented writer has claimed that the process is akin to reproducing a photograph in half-tone, which always has less snap than the original because it is made up of lots of tiny dots instead of continuous material. (Apparently he doesn't know that the original print is also made up of single grains of silver halides; what matters is that the dots be too small to see, and that they have adequate light-to-dark range.) It is true that when we reduce the entire sound field in a studio or hall to two voltages which vary with time, we are performing an abstraction which throws away a lot of information; and when we convert those voltages into a string of numbers we are performing another level of abstraction. It would seem on the face of it that there is no way to do this without losing information, and strictly speaking, this view is correct -- there is some information loss in the a/d and d/a conversions. Furthermore, there is no reason to think that these conversions are now being made as well as they possibly can be. But the idea of making the conversion, and the basic method by which it is being done in commercially available PCM gear, have been around for a long time, just waiting for the hardware with which to do the job. So we have a somewhat unusual situation technically, in that quite soon after it became possible to do the task, we are doing it very well. So well, in fact, that the losses inherent in the additional level of abstraction are far less than the losses in the best analog storage/retrieval systems. So: digital tape sounds better, sometimes much better, than analog.

Most of the criticism of "digital" has been in response to analog discs made from digital tapes. The disc cutting, plating, pressing, and playback processes all have worse problems than the digital tape recorder is likely to have, so such opinions contain little or no real information. What these reviewers seem mostly to be reacting to are the vagaries of performance, hall, microphones, miking, and mixing, which are now being revealed with significantly greater clarity. With both tape recording and discs improving simultaneously, we really are hearing more than we used to of what the engineer did.

I have heard the output of other digital systems, including tape recorders by Soundstream, 3M, Sony (PCM-1600), and Mitsubishi, and PCM disc systems by Pioneer (Philips optical) and JVC. My reaction has generally been that I was listening to the outputs of various microphones and mixing boards, with very little added or taken away. Of course, I have no firmer ground for this conclusion than anyone else might have for ascribing the flaws in the various sounds I heard to the digital process. Any of these professional systems could have had (just to pick an obvious example) audible frequency response errors due to design errors or parts tolerances in the anti-aliasing filters, and I would have no way of knowing it. I _have_ seen a high-resolution frequency response plot of a Sony PCM-1600, and it was even flatter than the PCM-1 tested here.

In the end, the _only_ test that gives any real information about the system itself is to compare the input with the output in a straight-wire bypass test, preferably with live material of wide bandwidth and dynamic range. With so much being said about the need for 18-bit systems and sampling rates of 100 to 200 kHz, the PCM-1, with its 13-bit words and 44 kHz sampling seems an unlikely candidate for perfection. But if you get hold of one, and do the test, I think you'll agree it comes very, very close.