RECORDING
ENGINEER/PRODUCER

Digital Price Breakthrough —

THE dbx MODEL 700 DIGITAL AUDIO PROCESSOR

Utilizing Companded Predictive Delta Modulation

— DESIGN PARAMETERS AND SYSTEMS IMPLEMENTATION

by Robert W. Adams, dbx, Inc.

— OPERATIONAL ASSESSMENT AT CRESCENDO STUDIOS

by William Ray, Crescendo president

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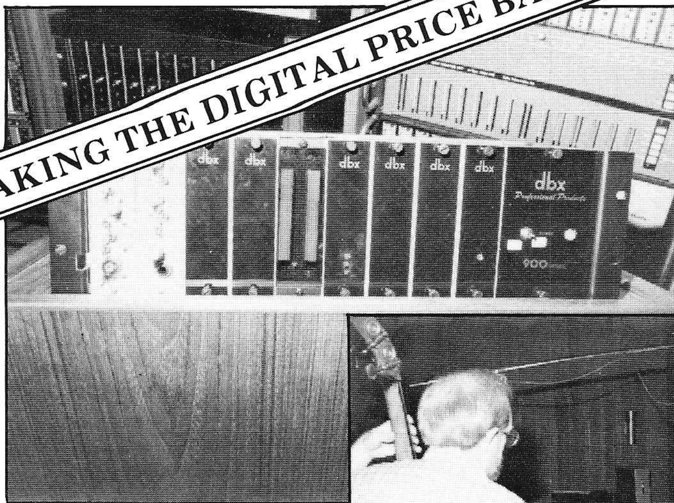
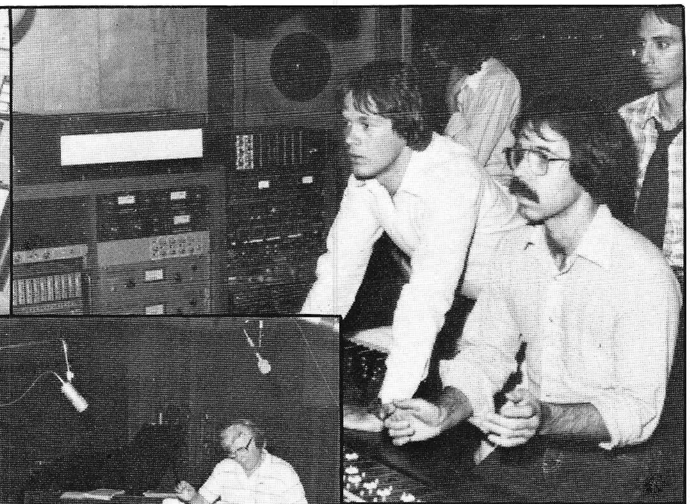
BREAKING THE DIGITAL PRICE BARRIER



Prototype Model 700



Trials at Crescendo

The dbx Model 700 Digital Audio Processor

Design Parameters and Systems Implementation

by Robert W. Adams
Senior Project Engineer, dbx, Inc.

Just about everyone who has heard an original digital recording has been impressed; most are enthusiastic. The virtual absence of distortion, noise, and wow/flutter makes the sound far superior to that of analog. But, because of the tremendous cost involved, owning a digital recorder is not exactly commonplace in the world of professional audio. For the "semi-pro" studio and serious recording musician, owning a professional quality digital recorder is an impossible dream. Although the cost of these digital machines will fall somewhat over the years, their complexity will make them more expensive than analog recorders for some time to come.

Apart from the expense of their respective recording equipment, there is a gulf separating the digital and analog engineer. The former lives in a world of numbers, bits and bytes, while the latter is more at ease with the use of one op-amp and a handful of resistors and capacitors in a tone-control circuit, than with incorporating 30 digital ICs.

Since dbx has considerable experience in analog audio R&D, it was decided to take advantage of the best of both worlds. By combining analog techniques with available digital technology that had never before been applied to the recording process, digital sound could be made affordable to every studio. The result is the dbx Model 700 Digital Audio Processor.

Evolution of the Model 700

The first goal in the design process was to find a form of analog-to-digital and D/A conversion that was both high-quality, and inexpensive. Ruled out because of their high cost were 16-bit linear PCM converters; 14-bit processors were somewhat cheaper, but didn't have the dynamic range needed

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Operational Assessment at Crescendo Recorders, Atlanta

by William Ray
Crescendo President

Ever since the invention of the audio tape recorder, deficiencies of magnetic tape as the storage medium have been a major stumbling block in the recording industry's never-ending quest to perfectly reproduce an audio signal. To be a little more specific, tape hiss, print through, limited dynamic range, high-frequency dropouts, head bumps and other frequency response related problems in the past, have seemed like insurmountable problems. Over the years, however, one by one these problems more or less have been dealt with.

Given that ours is an industry staffed by creative engineering types, it's hardly a surprise that so many products and ideas have materialized to deal with the limitations of the magnetic tape medium. One of these "creative engineers" whose work has enabled us to scale the insurmountable "Mount Tape Hiss" located at the beginning of the "limited dynamic range" of mountains is David Blackmer, president of dbx, Inc. Blackmer's development of the voltage-controlled amplifier forms the heart of dbx noise reduction, which provides the reduced tape hiss and enhanced dynamic range we've been looking for (on paper anyway). But, it's still a Band-Aid solution; most of the problems associated with the tape and the format are still there.

The Digital Answer?

Now, this brings us to "Digital Audio." Surprised? I was. You may ask, as I did, what does dbx have to do with digital audio? Read on.

Crescendo Recorders (formerly the Sound Pit) is a multi-24-track recording complex with a complete in-house video post-

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for professional use. Adaptive Delta Modulation (ADM) was attractive because of its cost, but after critical listening to material of very wide dynamic range, it was felt that the overall sound was not good enough for digital-recording applications.

After months of study and deliberation, a system was conceived and devised that offered several improvements in audio performance over ADM. This system was dubbed "Companded Linear-Predictive Delta Modulation," and will be described in detail shortly. The results of listening tests over this system were very encouraging, and convinced us that we had found a low-cost alternative to 16-bit PCM for professional digital recording.

Next we had to choose the storage medium. As is well known, the bandwidth requirements of digital recording are much higher than can be accommodated on an analog tape recorder. The design of a special set of tape heads to be used on a conventional transport was considered, but we decided that this would be too expensive, and take too long to implement. Finally we settled on videocassette recorders, which have adequate bandwidth, are readily available in several formats, and are produced in sufficient quantity to be comparatively inexpensive.

After these decisions, the first prototype was built. Initially no error correction was used because we found that our method of A/D conversion was fairly insensitive to bit errors. In fact, during normal program material, errors of up to 50 bits frequently were inaudible. But we also found that the largest of the dropouts on video tape would indeed cause clicks to be heard during low-level passages. Thus the next prototype was built with full digital error correction. Although this additional circuitry increased the cost, the unit could still be priced far below competing 16-bit PCM systems. This second prototype was used to record a wide variety of instruments and musical materials, both in studios and in concerts. It passed all tests with flying colors.

A/D Conversion: Companded Predictive Delta Modulation

Delta Modulation has been used for years as a low-cost means of A/D conversion. In this digital process, the numbers derived in the A/D represent *differences* between sampled voltages, rather than the instantaneous voltage produced in a "conventional" PCM audio processor. ("Delta" is the mathematical term for change or difference.)

Because it is based on changes in level, rather than absolute values, the dynamic range of Delta Modulation is restricted at the loud-end by slew-rate limitations — the signal slope becomes too steep for the A/D to track — and at the soft-end by the familiar quantization noise inherent in all digital recording systems. At high frequencies the dynamic range is especially limited, but even at lower frequencies it is not sufficient for serious audio applications. To extend Delta Modulation's dynamic range, Adaptive Delta Modulation (ADM) adjusts the step size to suit the dynamics of the input signal.

The analog-to-digital conversion process in the Model 700 differs from that used in normal ADM in two important respects. First, rather than vary the step size to follow the signal, in the dbx converter the signal is varied with a voltage-controlled amplifier to avoid overloading the fixed Delta Modulator. Second, to lower the quantization noise, the fixed Delta Modulator uses a "linear-prediction filter," which relies on the history of the audio signal to predict its future. These two differences between AMD and CPDM result in substantial performance improvements. To demonstrate, we have to go into detail. First, let's look at the high-precision compander (compressor-expander) used in the Model 700:

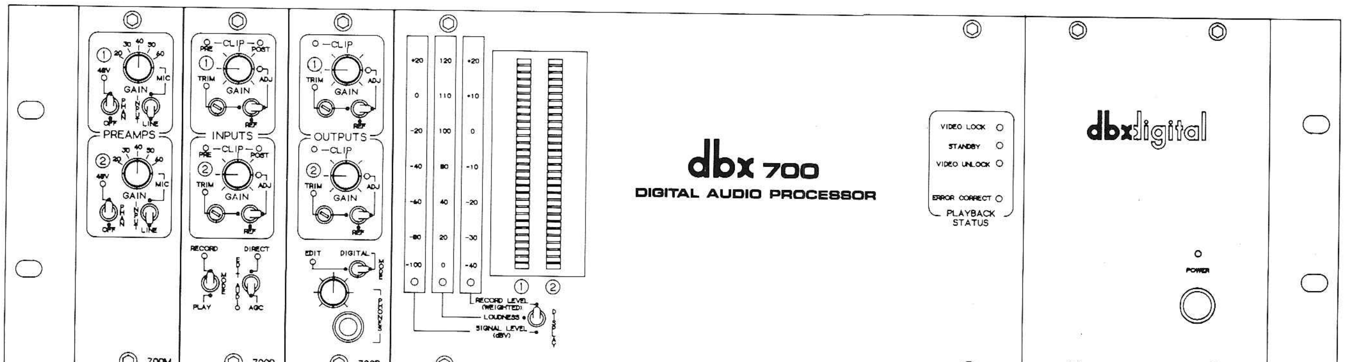
• *Companding versus Adaptive.* In ADM, step size is varied according to the average slew rate (speed of change of the input signal). A burst of high-frequency, high-level input signal requires a large step size, so that slew-rate limiting can be avoided. The problem with doing this, however, is that the range of practical adjustment of step size is limited to around 500:1, and at the smallest step sizes the comparator may not operate ideally, or even close to it. Also, the lack of dither noise can result in the noise floor being non-white (equal intensity for all frequencies), and signal-dependent.

The dbx system overcomes these problems by using a VCA in front of a fixed, non-adaptive Delta Modulator (Figure 1). When a large signal with a high slew rate is present, VCA gain is reduced, which lowers the slew rate of the signal passed on to the Delta Modulator. Thus, the input is adapted to the fixed step size of the Delta Modulator, rather than *vice-versa*. In playback, signals are decoded complementarily: the output of the fixed Delta Modulator is applied to a VCA whose gain is the exact inverse of the encoder's VCA gain.

The range of gain available from the VCA is beyond 120 dB, or voltage ratios of more than a million-to-one, which is a great improvement on the range available from ADM. Furthermore, using the fixed-step-size Delta Modulator lets the comparator have enough signal to operate properly, which also increases the available dynamic range. Finally, dither noise can now be added at the input to the fixed Delta Modulator, to eliminate any noise-floor anomalies ("birdies" and other such tonal effects) that are possible with ADM.

The signal that controls the gain of the VCA comes from a sophisticated level-sensing circuit that uses information present in the Delta Modulator's digital output. Being quite complex, this circuit cannot be fully explained in the space available here. Suffice to say that the VCA gain now can change very quickly to follow musical transients, but will change slowly for material that has slower dynamics.

It should be noted again that this level-sensing circuit obtains its information directly from the bit stream in both encode (record) and decode (play). Since these bit streams are identical in each case, mistracking (non-complementary VCA gains) *cannot*



occur.

- *Linear Prediction.* One of the problems affecting both ADM and companded DM systems is that the noise floor can change with signal level. This occurs because the step size is changing to follow the input, and step size is what determines the level of quantization noise. Generally, if the changing noise floor is far enough below the signal, its modulations are inaudible. Linear prediction is a method of increasing the dynamic range of a fixed Delta Modulator by more than 10 dB, and this increase is sufficient to eliminate any possibility of hearing noise modulation.

By way of illustration, let us assume a situation where the Delta Modulator has a fixed step size of 10 millivolts. Therefore, if its last "guess" at the input level was too high, the next will be 10 mV lower. Now, let us assume that of the last 10 guesses about signal voltages seven were too low, and three were too high. We might reasonably infer that the signal level was increasing. We could then shift the step sizes from ±10 to, say, +15, -5 millivolt, which is in line with our expectation (based on the recent history of the signal's behavior) that the signal is more likely to change in a positive than a negative direction. Note that doing this does not change or lower quantization noise: the difference is still 20 millivolts between +10 and -10, and +15 and -5 mV. But it *does* increase the maximum slope (steepness, or slewing, or speed of change) that the modulator can follow without slew-rate limiting. Hence dynamic range is increased, as well.

In practice, this alteration in the balance of "plus" and "minus" step sizes is achieved by a "linear-prediction filter." This filter is substituted for the simpler filter (integrator) normally found in a Delta Modulator, and is designed for maximum dynamic range. A comparison between linear-PCM converters and the dbx Model 700 system is provided in Table 1.

Memory

The dbx 700's memory has 16k bits of random-access memory storage for wow/flutter absorption, data interleaving and de-interleaving, and video requirements. During recordings, the A/D converter produces a steady stream of bits. The video format, however, has several areas where data can't be recorded (described below), so the memory is asked to store the data bits during these times. Of the 16k memory, 8k is for data interleaving (time scrambling), and 4k for storing data during the video-sync intervals.

During playback, the memory must supply the D/A converter with a steady stream of data while receiving the data from the VCR. But the VCR introduces wow and flutter, which makes the bit rate sent to memory variable. The memory absorbs these variations with the last 4k bits of storage; this results in a very low flutter in the decoded signal (less than 0.01%).

Error Correction

The dbx A/D conversion method, unlike linear PCM, is inherently tolerant of errors. In linear PCM, single-bit error may cause the most significant bit (MSB) to be in error. This MSB error might produce a disastrous full-scale spike in the audio output.

In the dbx Model 700, there's no such thing as an MSB; all the bits have a value just large enough to keep up with the signal's sample differences. For this reason, errors of 30 bits or less are usually inaudible during normal program material played over the dbx system. Professional U-Matic VCRs typically have very low bit-error rates, due to the high quality of the tape used, and the greater head-to-tape velocity; dropouts greater than 300 bits are quite rare. Consumer VCR formats often have longer dropouts, up to about 600 bits.

While it is recommended that the dbx 700 be used with a U-Matic-type machine, Beta and VHS units, being less expensive and offering longer recording time, may be used in situations where economizing is called for.

The dbx error-correction circuitry works by adding one extra parity bit for every three data bits. The parity bits are mathematically derived from the data bits, so that any bit errors on playback will produce a unique error pattern in the received parity bits. This error pattern is decoded to find exactly which bits are in error, and the offending bits then corrected. This correction circuit works in conjunction with the memory interleaving in such a way that a long burst error is presented to it as a series of short errors separated by good data.

Video-Format Encoder and Decoder

The format generator, or encoder, produces all the necessary synchronization, blanking, and equalizing pulse signals required to make the digitized audio signal look like the standard NTSC video signal, and thus acceptable to the VCR. It also controls the memory so that data bits are recorded only in the allowable video intervals.

The dbx video format records 128 bits per horizontal scan line and uses 224 lines per video field (out of a possible 262.5, the NTSC standard). The remaining lines are left blank to allow for the video-synch interval, for the special timecodes used for editing, and for the synchronization of several VCRs.

The decoder extracts the data from the video waveform on playback, and writes them into the memory. To do this, it must separate out the synch and data information, and decide which horizontal lines contain valid data. Unusually extensive protection is employed so that VCR noise and tape dropouts, which can easily look like valid synchronization signals, don't fool the processor.

Analog Display and Control Functions

Extensive metering facilities provide information about both the dynamic range and level of the input signal. The display is a column readout with 30 LEDs for each of the two channels. A peak hold with slow decay is also incorporated. The display can serve three selectable functions:

a) *Record-Level Indicator.* This has a range of 60 dB (2 dB per LED) and is pre-emphasized to follow the headroom characteristics of the A/D converter. Brief transients that exceed the maximum record level (+20 dB) will not clip because of the transient-speedup circuit in the level detector. Continuous operation above the maximum indicated

TABLE 1: A Comparison between Linear-PCM Converters and the dbx Companded Predictive Delta Modulation

	16-bit Linear PCM	dbx System
Cost	Very High	Low
Dynamic Range	90 dB	More than 110 dB
Sensitivity To Bit Errors	High	Low
Bit Rate	Approximately 770k per second, plus error-correction overhead	Approximately 700k per second, plus error-correction overhead
Anti-Aliasing Filters	Complex, hard to build, large phase shifts	Simple; small phase shifts
Maximum Level Flat With Frequency?	Yes	No
Distortion	Low	Low
Frequency Response	Depends on anti-aliasing filters; usually very good	Very good

record level is not recommended, however.

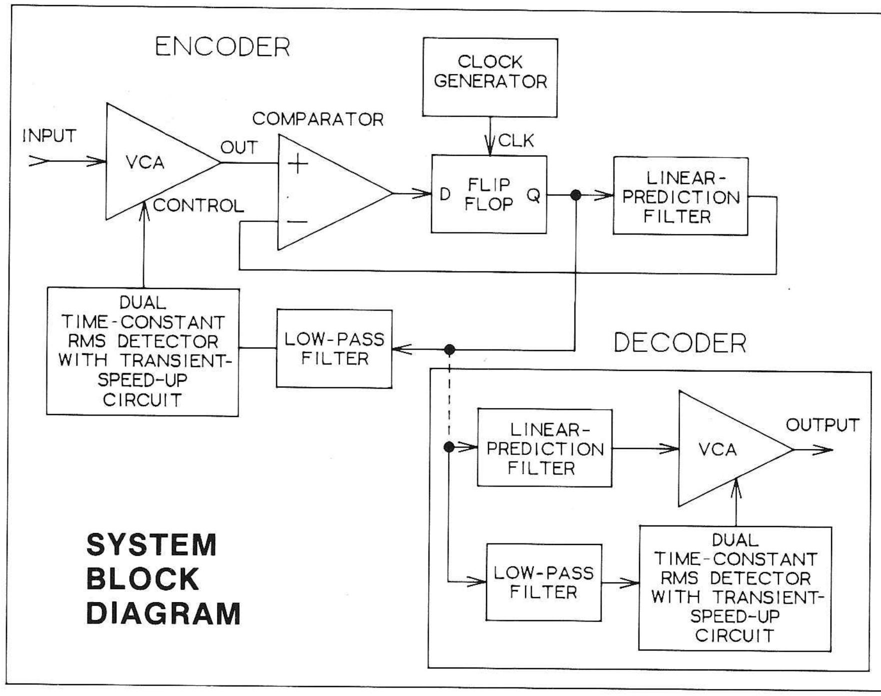
b) *Signal-Level Meter.* This is an RMS-responding, non-weighted indicator with a total range of 120 dB (+20 dBV to -100 dBV). In record mode, this meter reads the line input or output of the mike pre-amp, if one is used. In play mode, it reads the unit's line output level.

c) *Loudness Meter.* This incorporates complex dynamic-filtering circuitry that simulates the equal-loudness contours of the ear. It follows the Stevens curves (the modern version of the old Fletcher-Munson curves) to within 2 dB over the entire 120 dB range of the meter. This feature is invaluable in making dynamic-range measurements, where a "flat" meter often will give too high a reading because of low-frequency room noise. The inputs to the loudness meter are switched in the same manner as the signal-level meter. Sensitivity of the microphone is used to set the reference level.

The analog-input circuitry contains all necessary control functions. A three-position switch selects a front-panel level-control pot; a trim pot adjustable through a hole in the front extrusion; or an internal non-adjustable reference level. This last position is provided so that the unit can become a unity-gain device from record to play, which makes it easy to play back a recording at the same sound-pressure level as the original, if the sensitivity of the microphone is known.

Clipping LEDs are provided both before and after the level-control stage. In a device with such a large dynamic range, the gain structure is quite important; if the front-panel LED is set too low, for example, dynamic range may be lost.

The analog-input section also provides a signal for recording on the VCR's audio tracks for use during editing. This is necessary because the digital audio information cannot be recovered when a VCR is put into slow-motion to search for an edit point. A 2:1 compressor may be switched in so that wide-range material can be successfully captured on the VCR audio tracks.



SYSTEM BLOCK DIAGRAM

The analog-output section contains two output buffers capable of driving 600-ohm loads to +24 dBm; a stereo headphone driver; clip LEDs; and another three-position switch to select among front-panel pots, screwdriver-accessible trim pots, and an internal non-adjustable reference calibration. Low-noise circuitry is used throughout, and all electrolytic capacitors in the signal path are paralleled with smaller, non-electrolytic caps for good audio quality. Electronically balanced outputs are standard, and may be defeated if unbalanced ones are desired.

An optional low-noise mike pre-amp module can be plugged into the last slot in the frame. Each channel has controls for gain (20, 30, 40, 50, and 60 dB), 48 volt phantom powering (on/off), and Line/Mic source select. Our low-noise circuitry adds less than 1 dB of noise for microphone impedances from 100 to 1k ohms.

Construction

The dbx 700 is completely modular, all circuitry being contained on 10 printed circuit boards that plug into a back-plane. The complete power supply, including transformer and AC input, is also modular and plugs into the back-plane. High-quality XLR connectors are used on the rear panel for all audio input and output connections, and BNCs for connections to and from the VCR.

* * *

It was through the marriage of analog and digital design that dbx hoped to spread the benefits of digital sound among those to whom they might otherwise have been delayed. For the first time, a studio owner can purchase a digital recording system — the dbx 700 Digital Audio Processor and a professional-quality VCR — at a price comparable to that of a good two-track analog recorder. This feat was accomplished by innovative circuit design, which is most apparent in our A/D converter, wherein a unique combination of analog and digital technology provides extremely high performance at remarkably low cost.

The main goal in designing the dbx 700 was to lower the cost of digital processing sharply in order to bring digital-recording capability to every engineer and studio that could afford a top-quality analog recorder, not to mention the associated processing equipment. The low (under $5,000) price tag meets this goal. We also believe that the dbx 700 sounds as good as, if not better than, the finest digital equipment currently available from the major manufacturers. Delivery is targeted for next summer. ■■■

AN OPERATIONAL ASSESSMENT *by William Ray*

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production facility. Our clientele is diversified, and has included such acts as Ted Nugent, Cheap Trick, Lynyrd Skynyrd, and Kansas. As with most studios of our size, the ability to attract new clients — as well as keeping old ones — depends upon our ability to maintain a "state-of-the-art" facility. With microprocessors raining down on the public from every direction, "Digital Audio" has finally entrenched itself in our clients' vocabulary, and has become one of the more popular buzz words. The pressure had been mounting for us to commit to a digital mastering machine.

Normally, I would not question the validity of a particular piece of equipment that so many clients had requested. (Heaven help the studio that interferes with its clients' creative pursuits!) However, we had definitely been

OPERATIONAL ASSESSMENT AT CRESCENDO RECORDERS

procrastinating on this purchase. My reasons for procrastination centered around several very fundamental concerns.

First, was format. Mother technology has been *very* generous in recent years, by bestowing upon our industry many technological breakthroughs. However, man's nature being what it is, we have managed to significantly slow the implementation of many of these technological breakthroughs by spending years on debating the best way to proceed. A classic example of this phenomenon is EQ and alignment for analog audio tapes. We've had to deal with NAB and IEC for years, and only recently was a "standard" agreed upon. There is still no recognized standard tones or levels on a master tape for set-up and alignment purposes. (I find it ironic that a resolution will finally be made at the same time that we discover a technology to make analog tape obsolete as a format.)

As a studio owner preparing to make a significant investment ($30,000+), I have to ask some very important questions concerning the establishment of a new recording format:

- What will the storage medium be? (The options now are audio tape, and video tape.)
- If the storage medium is video tape, will it be ¾-inch or ½-inch — Beta or VHS?
- If the standard is video tape, how will I edit? Will I be able to use a low-cost video editor?
- What will be the standard format for analog to digital conversion? PCM, ADM, or something entirely new?
- Assuming we can standardize on an A-to-D format, what will be the sampling rate, since this has a significant effect on signal quality. If it is too low, it causes significant technical problems as it is increased (especially with PCM A-to-D).
- Will the cutting facilities I use be able to decode my digital masters?

Until some effort is made to answer these questions, purchasing a digital recorder is kind of a "pig in a poke." A $30,000 or more investment (over $60,000 for the Sony PCM-1610 processor with its editor) could be a complete loss in a year if a non-compatible standard were to be adopted. Despite digital's obvious audio attributes, the lack of economic security so far has kept us (and many others, I'm sure) from purchasing a machine.

My second concern is price. I've seen the price of all digital related technology come down drastically in recent years; studio-quality digital delay lines, for example, have gone from $3,000 to $499.00. I assumed (correctly) that digital audio recorders would follow suit.

Despite attempts to enlighten my clientele to these problems, requests for digital audio have continued relentlessly. We were losing the battle. I agreed to appropriate funds for a digital recorder, and began researching.

It looked as though the storage format would be ¾-inch video for two-track masters. A few phone calls proved that although *none* of the mastering facilities we used had digital replay machines, they did have access to a Sony PCM-1610 and video recorders on a rental basis of $500.00 per day — one-day minimum — when available. We were told it would cost us $29,500 for the Sony PCM-1610 processor. In addition, we would have to spend between $4,000 and $8,000 for a ¾-inch U-Matic VCR.

The dbx Alternative

Randy Fuchs, my partner and fellow owner of Crescendo Recorders, in a conversation with his long-standing friend, Lance Korthals, mentioned our decision to purchase the Sony system. Korthals, dbx pro sales director, felt that he had to let a good friend like Randy in on a "little secret." Well, his little secret may well be one of the most significant advances yet in our industry. As you must have guessed by now, dbx was developing a digital audio processor.

If it seems odd that dbx would enter into digital audio, think for a minute. This company has made one of most significant developments in reducing tape hiss and expanding dynamic range. Given that innovation, it has probably reached the limitations of analog audio. Where else could the company turn, but to digital?

In less than two weeks, the studio arranged for the prototype dbx 700 Digital Audio Processor, as well as a Sony digital machine and two ¾-inch U-Matic VCRs, to be installed in our Studio "A" for serious evaluation over a four-day closed session. (Special thanks to Tom Semmes and Associates for the loan of the Sony digital system.) Although this would be the first time we've had a digital recorder in our facility, I am no stranger to digital recorders. I have been to every AES and NAB show in recent years and, as I said before, we have been evaluating digital audio for quite some time. I am well aware of the attributes as well as the deficiencies of the different formats on the market, as well as fundamental A-to-D problems. I have to admit that Crescendo primarily was looking for potential problems or deficiencies in its evaluation.

Systems Evaluation

Our evalutation was set up as follows: We would eliminate the multitrack, and cut "live" straight to the mastering machines. This eliminated any analog tape link. Identical two-track mixes were fed to the Sony PCM-1610, the dbx 700, and to an analog Otari MTR-10, it being felt that the Otari represents the "state-of-the-art" of analog tape machines. The MTR-10 has adjustable phase compensation, and a unique head design that make it audibly superior to everything we've evaluated — in other words, the ideal "analog reference." All levels were calibrated for each machine's optimum performance. No signal processing was used, since limiters, gates, etc., would only mask deficiencies.

With the help of Dr. Robert Manchurian, a prominent Atlanta arranger-producer, and Albert Coleman, of the Atlanta symphony, Crescendo proceeded to book the most diverse and challenging sessions we could. These included a classical pianist, rock drummers, jazz percussionists, acappella vocalists, string sections and soloists, horn sections and soloists, plus jazz, fusion, and rock bands.

In light of the magnitude this evaluation was taking, it was decided to involve as many ears as possible. At dbx's request, we did not identify to anyone that the company's prototype was here. Our engineers, producers, and performing musicians listened to each cut, while the musicians auditioned only what they cut. After each cut, all three machines were played back, and simply identified as A, B and C.

Considering the diversity in listeners, I believe that we compiled some significant data. After all, who knows better what a violin should sound like? An engineer or the performing concert violinist? On the other hand, however, it's the well-tuned ears of an engineer that notices abrupt cut-off of long-fading resonance (due to error correction circuitry in some digital recorders).

When the results were in after an exhaustive four days, they were, to say the least, "interesting."

No one *ever* chose the analog recordings; the limited dynamic range was immediately apparent. The Consensus between the PCM-1610 versus the dbx 700 was split equally. Everyone agreed the difference was minimal. However, the more seasoned ears could ascertain between the two most of the time. There seemed to be no peer grouping as to preference. The engineers were split, but the musicians seemed slightly to prefer the sound of the dbx 700.

I have to admit in this "blindfold test" I did choose the Sony PCM-1610 most of the time. However, just when I though I could tell the difference, I chose the dbx 700, insisting it was the Sony. But my partner, Randy Fuchs, consistently picked the dbx unit as his preferred choice. Our engineers, Will Eggleston and Jim Boling, could identify which was which after about 20 seconds. They disagreed, however, as to which they liked better.

The slight differences in the two digital machines were most noticeable in the high-frequency transients. The Sony PCM-1610 seemed to be more "piercing," for lack of a better term. Depending on your perspective, our evaluators defined the Sony as harsh (bad) or brilliant (good). The dbx 700 was described by the same evaluators as slightly dull (bad) or smooth (good).

The noise floor was non-existent on both units (below the noise floor of our mikes and boards).

The low-frequency response was incredible on both machines. Low frequencies, I might add, are one area that analog machines can't touch digital — with or without signal processing.

There have been claims that PCM-based digital recorders have a tendency to chop off a signal that falls below a certain SPL, in much the way that a gate would. It is my understanding that error-correction circuitry is responsible for this. dbx informed us that its unit was *not* a PCM system, so we did listen for this anticipated problem. We were not able, however, to get either unit to "chop" any part of even the longest and softest fades.

Crescendo Recorders' co-owners Randy Fuchs (left) and William Ray during evaluation of prototype Model 700 against a "conventional" PCM digital audio processor.

Cost Advantage

One thing I've refrained from mentioning until now is the cost differential between the two digital recorders we listened to. The dbx 700 is priced between 1/6 and 1/7th the cost of the Sony PCM-1610. While the Sony is truly an excellent machine and certainly *cosmetically* much more impressive to look at, we are purchasing the dbx.

Performance-wise the two machines are on a par. There are some packaging features I think show excellent fore thought on dbx's part. They've constructed the unit in a lightweight, vertical rack package that is similar to their 900 Series modular signal processing frame. The new 700 system is modular, and gives the user the option of tailoring a unit for his particular needs. The available modules are input, output, and mike pre-amps, which permits a mastering facility to purchase a playback-only unit, for example. It also gives the "live" performance "direct-to-deck" user an extremely high-quality mike pre-amp. This pre-amp will be essential for esoteric digital recording, since most available mike pre-amps and consoles that have acceptable noise floors for analog use will not cut it for digital.

An interesting observation at this point is that what has been until now one of the quietest links in the audio recording chain will now be the noisiest — you guessed it, the microphone.

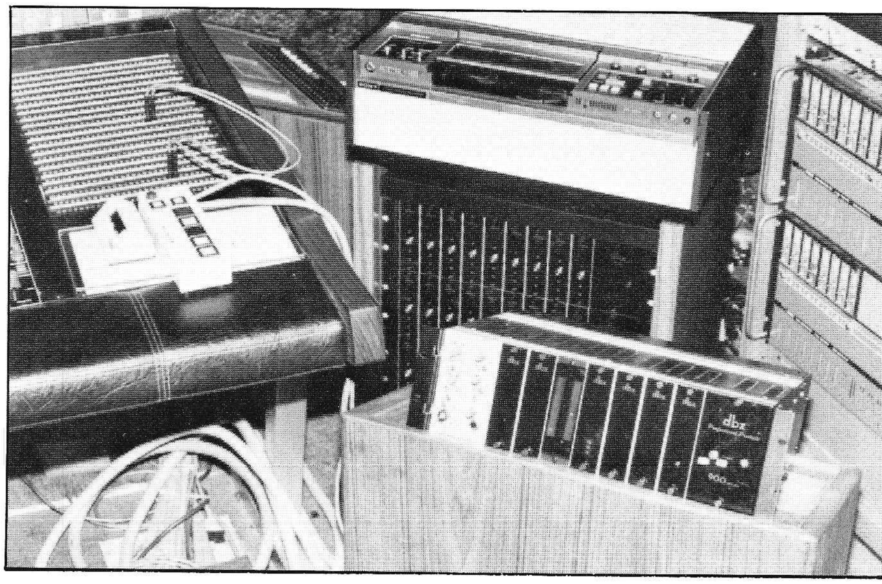
dbx has been successful in overcoming some of the objections (the biggest being cost) we've all heard about digital. However, there are a few problems that remain.

The dbx 700 uses a VCR and videotape. For editing, this means you either need two units and a video editor, or access to a video editor. However, on a positive note, any video editor that will interface with your VCR will suffice. While dbx recommends that it be used with a ¾-inch U-Matic, the 700 processor produces excellent results with ½-inch video tape recorders as well. The Sony and all others must use ¾-inch tapes. There is at least a $3,000 difference in the cost of a ¾-inch U-Matic and a ½-inch consumer VCR.

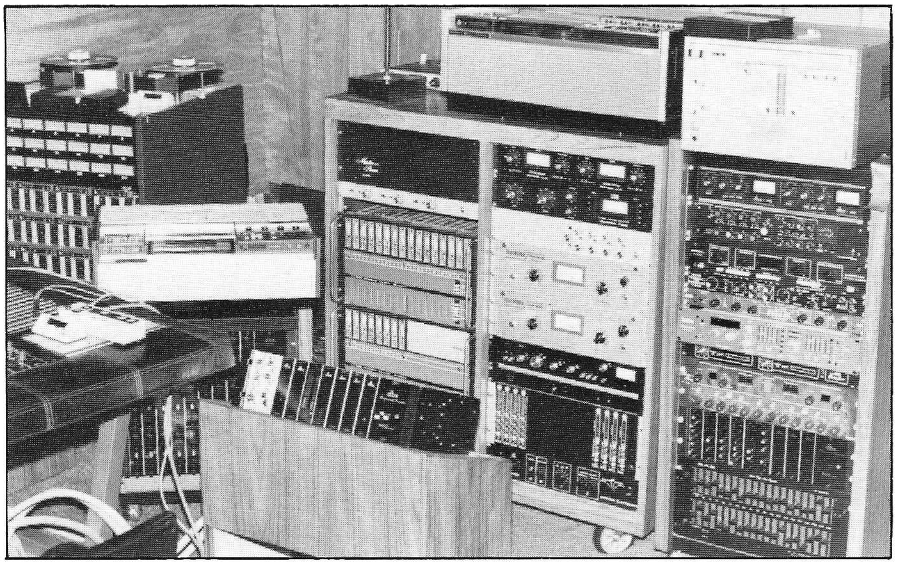
In our opinion, what dbx has accomplished with its digital audio recorder is certainly going to rock the industry. Facilities competing for album projects will certainly be forced to purchase a digital machine, or lose their business to the competition who has. Considering the cost of the dbx processor and ¾-inch VCR is roughly in line with a good analog recorder, price should certainly not be an obstacle.

Other users of high-quality half-tracks may be interested for other reasons. One very important issue that lies on the positive side of videocassettes is that as a storage medium they are very compact and easy to handle, and you don't have to worry about record/replay EQ, or tape speed.

Another plus with the dbx 700 unit is that a 60-minute ¾-inch U-Matic cassette costs $20.00 each in quantity. If you add up what 60 minutes of tape costs running at 30 IPS, you'll find yourself with four, 10-inch reels, or approximately three to four times the cost, with a considerable increase in



Close-up detail of prototype Model 700 and U-Matic ¾-inch videocassette recorder used to record digitally-encoded material.



The two digital audio processors — dbx Model 700 and Sony PCM-1610, plus companion U-Matics — used for comparison evaluations at Crescendo.

bulk. To users with extensive tape libraries — for example, radio broadcasters and radio post production — this alone could be reason enough to go to dbx's digital format.

Towards the Future

Before closing, and while I have the chance to "put it in print," I'd like to share some observations of the past and some projections for the future. As mentioned earlier, our inudstry has had to deal with a lack of standardization. Perhaps one very appropriate example to cite would be the Dolby and dbx noise reduction systems. Dr. Ray Dolby was first to come up with a system to significantly reduce the noise floor of a tape. However, dbx would soon be introducing an "alternative." And, as you all know, a triumphant victor did *not* emerge; our facility has both Dolby *and* dbx, and our clients swear by one or the other (or both).

In this case, had a format been established as a "standard" for noise reduction, we would have to give up audio integrity in some applications. Both systems have their attributes, as well as deficiencies.

As much as we'd all like to see standards set for a digital recording format, realistically I don't believe it will happen. Perhaps a by-product of "Yankee Ingenuity" is a common consensus that there is *always* a better way. This, coupled with healthy capitalist competition, will certainly lead innovative manufacturers, such as dbx, into alternative ways of manufacturing a digital recorder. The performance difference of going away from a PCM format, in the way that dbx has, is virtually beyond this listener's ability to perceive (hear). The cost advantage of going to dbx's encoding format is significant. The technology involved is simpler to execute than PCM, thereby enabling dbx to make significant reductions in component count, as well as size and weight.

Given that most studios probably do not have in-house personnel to repair digital recorders, I believe that dbx has a big advantage over its competition in that its new processor is less complex, and completely modular. With a few spare "cards," a studio should "theoretically" never have any downtime.

The dbx digital approach is, to our mind, certainly the most viable and well-thought-out yet. However, PCM-type recorders have already gained a viable foothold in our industry. Although current technology will not permit a PCM-based recorder to compete economically with dbx's approach, I think we will continue to see PCM-based recorders. And so — alas — we will, once again, have multiple formats, and no standardization. The only consolation may be that with the money we've saved on our recent purchase from dbx, I will be able to buy other innovative and new products. ■■■