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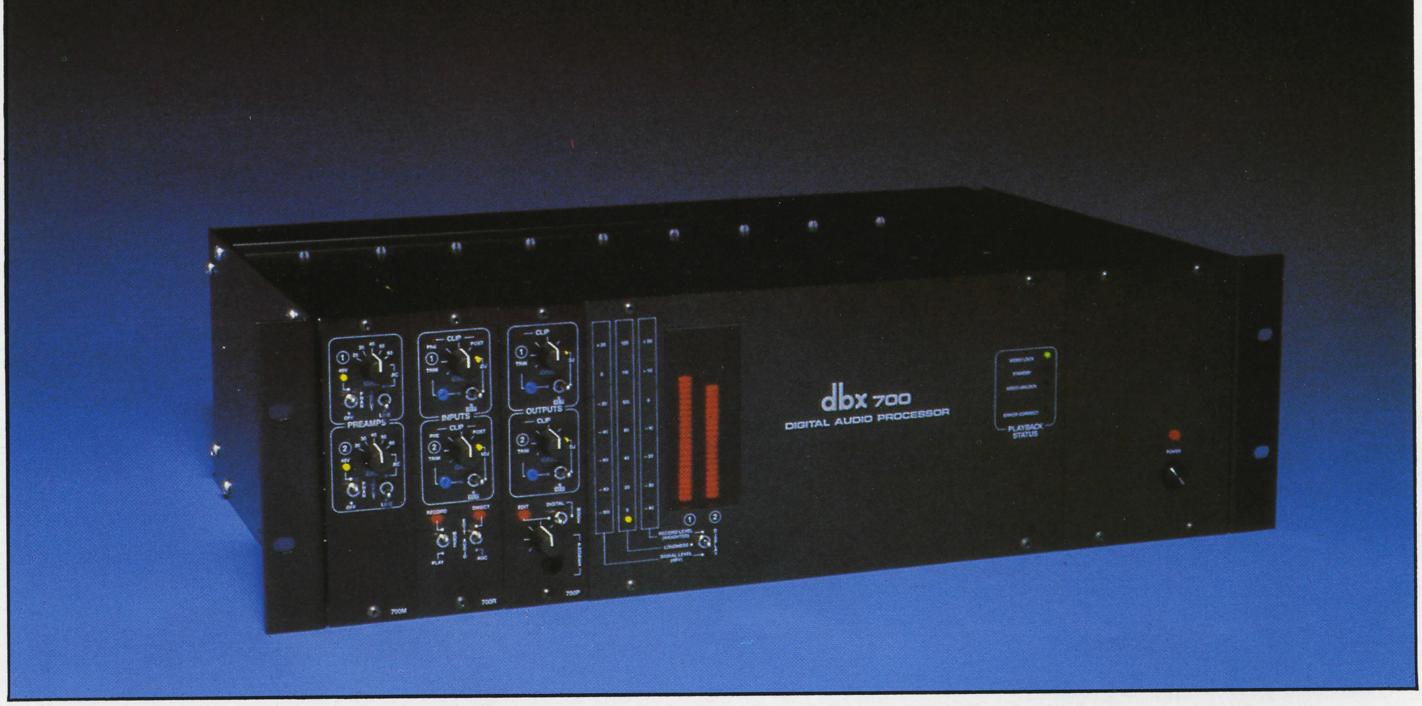
dbx digital–the full story

Noise reduction and gating

dbx Model 700 Digital Audio Processor



One of the most talked-about items of new hardware at the recent Anaheim AES Convention was dbx Inc's first foray into the world of digital recording. Their aim was to produce a professional 2-channel processor which would at least equal the quality of conventional linear 16-bit PCM systems, but at a fraction of the cost. The dbx *700* is the outcome of their research, and the low price—around \$5,000—is achieved by means of a novel approach to digital recording, known as CPDM: Companded Predictive Delta Modulation. In this interview, Robert W Adams, senior project engineer at dbx, discusses the development of the system and its future, including the projected development of a multitrack recorder with the same kind of price-tag per channel as an analogue machine.

''I started working on the project about two years or so ago,'' says Adams, ''after I had completed work on the *20/20* automatic equaliser. I was looking for a project which would top that! Basically, I came up with the method of conversion first; I had this idea of doing delta modulation—in which the numbers generated in the A/D represent *differences* between sampled voltages rather than voltages themselves—in a slightly different way, using linear prediction and so on, and I worked on that for a while, and got it to sound pretty good. Then I thought about what kind of products this would enable dbx to do, which we hadn't been able to do in the past.

''My view of PCM was that it made a lot of sense from a theoretical point of view. The math was easy, and it was all worked out. But it's a 'brute force' way to go analogue-to-digital, especially when you get into 16 bits, because of the accuracy required in the D/A converter, and the anti-aliasing filter requirement. I just had the suspicion that there was a 'sneaky way' around it, a better way that, psychoacoustically, would sound as good or better than what they're doing.'' Adams's view was that some of the novel approaches to linear PCM—like the oversampling technique used by Philips, in which 14-bit D/A converters with 16-bit accuracy are used—was not a solution to the anti-aliasing filter problem and was only really applicable to decoding because of the speed restrictions.

''So I thought, let's go for broke, and work on a digital audio recorder. There were a bunch of prototypes that worked to varying degrees. One of the things I thought about delta modulation was that it was very insensitive to bit errors. There is no such thing as a 'most-significant bit' anymore—it's a bit-stream, and all the bits have a value just large enough to handle the sample differences. Because of the video requirements we have a block format, but even that isn't theoretically necessary. Initially I thought we could get away without error-correction, because we had built dropout simulators and discovered that we could lose 50 or 100 bits and hardly notice the difference. The first prototype we carried around had no error correction, and that's the one we brought to the Boston Symphony Orchestra sessions in March 1982. It took over a year of work before we could build that prototype.

''We got back from the BSO sessions and the tapes sounded pretty good, but over time the tapes started degenerating, and you could hear the errors, so we knew that we needed error correction. We designed an error correction system and, once again, it isn't the standard Japanese system: we use a 'convolutional' method which is somewhat different.'' Conventional PCM has a defined word-length, and various mathematical operations can be performed on the data to obtain parity words which can be used to reconstruct the data to a certain extent, to re-create a missing data word, for example. For larger errors, a system of averaging between known samples is used. ''But,'' says Adams, ''there is some doubt as to how audible that interpolation is.

''We found that there are two types of error: tape dropout errors and tracking errors, where the VCR will just lose sync with the scanning. Even in consumer format, tape dropout errors are generally less than five TV lines. Tracking errors can be quite long, and exceed anyone's error-correcting capability, and the only way to tackle those is to dig into the VCR, use good tapes and use the right VCR. That's when the Sony system will go into its 'mute' mode. Incidentally, we

dbx digital—an overview

Richard Elen

recently tried our system on a new Sony *U-Matic* front-loader which incorporates 'video correlation' error correction, finding video noise by comparing between frames. This is a disaster for digital recording! Both the Sony system and our system would require that on that particular deck you take the side cover off and turn down the video correlation pot.''

Delta modulation

One approach is to use 'adaptive delta modulation', in which the step size—which determines how the system 'hunts', how large it can go from one sample to the next to follow the signal—is changed according to the signal. The larger the signal, the larger the step size is made to follow the signal. But when the step size is increased, the quantisation noise is increased too, so you end up with a form of noise modulation of the signal. This could be a problem. ''One of the problems we found with ADM,'' says Adams, ''is that the noise floor is not always 'white': in fact you can get little 'birdies' and 'tweety' noises which result from not using dither noise. The reason that dither noise is not used with ADM is that the step size, the quantisation level, is changing over a 1000:1 range. There is no single level of dither noise that will cover a couple of quantisation levels. What we used was a *fixed* delta modulator, and instead of adjusting the step size to suit the signal, we adjust the signal with a VCA (which allows an adjustment range of 1,000,000:1) to suit a fixed step size. That allows us to add a fixed, right amount of dither noise. Another difference that we have come up with is that our fixed delta modulator is not the simple integration type: it has a more complex filter than an integrator, and it accomplishes what is called 'linear prediction'. It looks over the past 20 or 30 samples of the signal. Let's say that of those samples, seven were too high, and three were too low, you would be able to tell that the signal was decreasing. So you would be able to predict, since music doesn't normally have large changes from one sample to the next (especially if you're sampling as we are at 700 kHz), what the next step size really ought to be.''

A bit rate of 700 kHz sounds very high, and it is tempting to compare that with linear PCM. At 48 kHz sampling rate, you could say that the bit rate is 48 × 16 = 768 kbits/s, but of course, the PCM system needs 16-bit absolute samples: the sampling rate (48 kHz) and the bit rate (768 kbits/s) are not the same. The dbx system does not use 'words': it is a bit-stream in which the sample and bit rates are the same. Such comparisons are thus rather misleading. The *sample* rate is the only real comparison that can be made. ''There is no way to come up with a 'word' format to describe the dbx system,'' says Adams.

In the dbx *700*, an analogue signal is applied to the input of an A/D (see **Fig 1a**). What happens after that? What comes out? ''It's a serial bit-stream. A regular delta modulator is a feedback system. The signal goes into one side of a comparator (**Fig 2**) and the estimated signal goes in the other side. Every sample period you compare the estimated signal to the real signal, and if it's too high, for example, you send a low signal out to an integrator, which simply ramps down. We have increased the degrees of freedom. Instead of just ramping down or up, say by ±10 mV as in a normal delta modulator, our system can shift the centre about which it can go down or up. It's not just zero: you can go +15 mV and —5; or +30 and +10 or whatever. The quantisation—which is the difference between those two steps, the maximum error that you can have—stays the same. But the maximum *slope* that you can track goes up. This is all by looking at the past history of the signal—the 'predictive' part of CPDM.'' But surely this will create ringing with a square wave, where the system is estimating a rising signal, and then suddenly it stops rising? ''It will overshoot a little,'' says Adams, ''but that will give you ringing at an ultrasonic frequency, well above 20 kHz. And it doesn't ring very much. We're not looking at the past history in terms of milliseconds: it's a matter of microseconds.''

Presumably, this means that there is no real necessity to place a solid limit on frequency response, unlike PCM systems. Says Adams: ''There's no brick wall. It's more 'rubbery'!'' There is, of course, a theoretical limit at 350 kHz, half the sampling rate, but there isn't much headroom up there! ''My own feeling on frequency response,'' says Adams, ''is that a response beyond 20 kHz is not necessary or audible. On the other hand, the problem with the anti-aliasing filters that we're seeing is that in order to get the high rate of rolloff, the phase shift *in* the band is extremely high. There is a lot of controversy about whether that's audible or not. With 10- or 12-pole filters, you're talking about errors of thousands of degrees at 20 kHz. And that extends all the way down. The area where there's likely to be trouble is in the mid-band region, where phase distortion is more audible. Because our sampling rate is so high, our anti-aliasing filter can be very gradual, reaching —60 dB at 200 kHz. This results in a phase shift of less than 100 degrees at 20 kHz.''

FIG 1a. dbx DIGITAL ENCODE

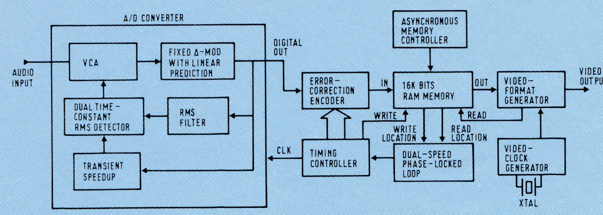


FIG 1b dbx DIGITAL DECODE

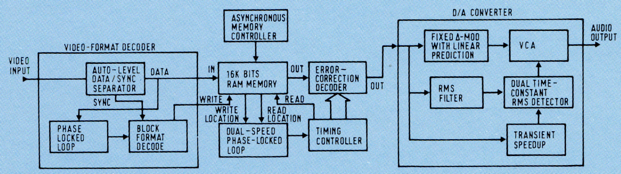
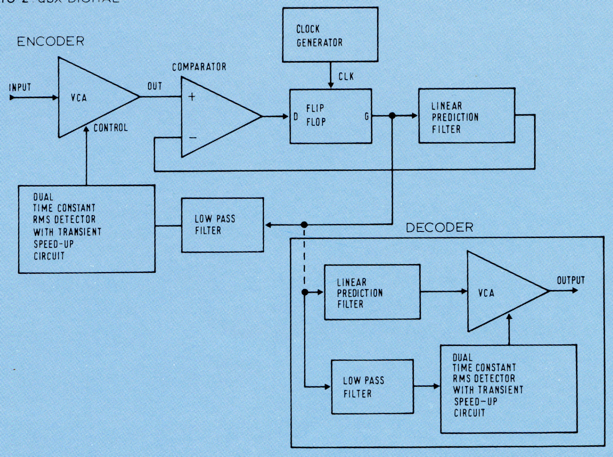


FIG 2 dbx DIGITAL



Compansion and prediction

On the dynamic range front, this linear prediction gives distinct advantages. Without linear prediction, you would be looking at about 55 dB signal/noise. ''With linear prediction,'' says Adams, ''we get that up to about 70 dB. The system is rather better at making that 'next guess' than a fixed delta modulator would be. Then we have a compander, which is quite different from those used on our analogue tape systems. The RMS detector, and everything that controls the gain of the VCA, comes from the digital output (see **Fig 1a**). One interesting thing about the output from a delta modulator is that the signal itself—in terms of frequency-domain analysis—is present in the bit-stream. If you put the bit-stream into a Fourier analysis machine, you will see the analogue signal right on it. So right from this bit-stream, we can look at how hard the delta modulator is being pressed, and adjust the VCA gain accordingly. The whole key is to make the speed of the gain-change similar to the speed of the music. For slow, non-dynamic music you want a nice, heavily-filtered control voltage. For fast stuff, you may want to make it go as fast as possible. So we have an RMS detector which has basically two speeds, slow and fast. On top of that we have the transient speedup circuit which senses a signal overload lasting more than about 20 μs, at which point it will make the RMS detector go really quickly. This, of course, would not be possible around an analogue tape recorder, because you just couldn't decode it at the other end.'' Of course, as the dbx system gets its control voltage from the bit-stream, which is the same at 'both ends' of the system, there is absolutely no possibility of the compander mistracking.

The compansion system works more or less linearly up to a certain point, above which it turns into an infinite ratio when the transient speedup circuit comes in. This means that you can overload the system and still recover it at the other end. Although the expander at the other end goes into an 'infinite' mode for a brief period at such times (not a situation to happen for long),

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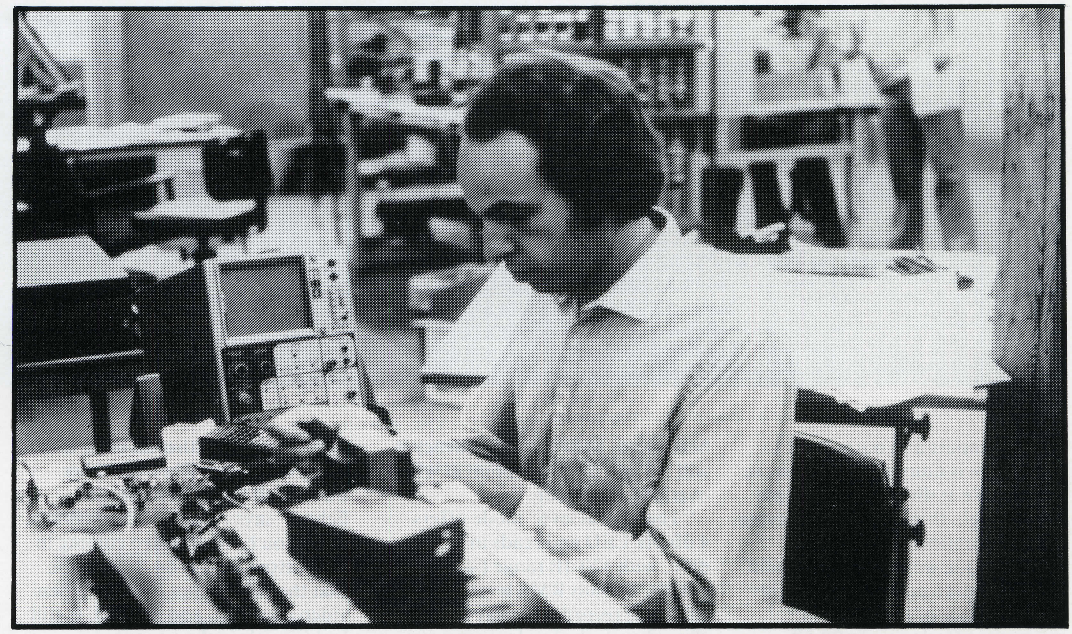
it does mean that the system can cope with, say, a hot snare drum beat without shutting down or clipping, as would a PCM system. This makes the system more tolerant of operator error. The actual compansion curve has no one ratio: it has a number of carefully-designed 'knees' in the curve, tailored for best results. Dither noise is intentionally built into the compander circuitry, just as PCM systems often have noise designed into the filters for the same purpose.

The bit-stream next passes to a set of circuitry which clocks the bits into and out of 16K of 70 ns, H-MOS static memory, to even out wow and flutter, etc and is then coded into video format for recording on a VCR. "There's a dual-speed phase-locked loop there. The whole point about getting rid of wow and flutter is that you have a memory that is large enough to absorb the variations. You feed the bits in one side and feed them out at a constant rate the other side. But you need some kind of timebase corrector, a phase-locked loop, to ensure that the average 'in' rate is the same as the average 'out' rate, so that you neither overflow nor run out of data. It has to be a very slow PLL so that the wow and flutter spectrum is heavily filtered, so there could be difficulties getting into lock when you first start up. This was one of our early problems. So we went to a dual-speed PLL, which senses when the input and output rates don't match, and switches into its faster rate, settling in about a second or so. After about five seconds it switches into the slower mode, where it filters wow and flutter. For the first two seconds after turn-on, you might have the wow and flutter of the VCR in the output." There is only about a 1.2 μs delay between the two channels, which are alternately sampled, unlike PCM in which there might be as much as 20 μs delay between channels, and that appears to be at the threshold of the ear's ability to detect phase differences.



Video formatting

"The video clock-generator is crystal-controlled. The system uses standard NTSC video format, recording 128 bits/line, on 224 lines out of the possible 262.5. So you leave a bunch of lines in the middle blank for putting in vertical-interval time codes and so on. Using VCRs there's always a point where the head switches, and this is designed always to be in the vertical interval—you can't ever record over that!" There will be a PAL version very shortly. "Another part of the video format is that we have a 'white reference pulse' stuck at the beginning of the line," Adams continues. "This is higher in level than the data pulses, and this gets round the problem with VCR AGC circuits, which often have differential nonlinearities, and a frequency response which worsens near the white end of the scale. As you need as much bandwidth as possible, you need to avoid recording in the 'whitest' end of the video scale. The white reference pulse stops the AGC from recording the data at the highest possible level." The reference is stated at the beginning of each line, and must be wide enough to 'set up' the AGC correctly. It is in fact three data pulses wide. "We also use the reference pulse to identify which lines have data on them. They don't appear during the vertical interval. One main problem we had in this area was that while the system could allow for a missing reference pulse after it was running, what would happen if it missed the first reference pulse after a vertical blanking interval? This takes a good deal of fancy decoding in the block decoder. We look at pulses during the vertical interval and if a reference pulse isn't recognised after a certain time period, we give up on that line—the error correction will deal with it anyway—and we send a signal to the memory telling it not to shift the next set of data into memory starting from 00; go to position 128 and start from there. As long as you don't lose sync, you're all right. Before the data comes along, there are two horizontal lines' worth of sync information for the PLL to lock up."

The signal comes off the tape and through an automatic level-sensing data separator. The self-same circuits are used for encode and decode, the unit not being a simultaneous encoder/decoder. The clock is regenerated via a PLL, the video block format is decoded, and the reconstituted bit-stream is written to memory. "What's necessary," says Adams, "is that every bit that came from a certain place in memory and on to the video tape, has to come back and get to the *same* place in memory. That's what is necessary for de-interleaving. The beginning of every frame starts at a particular place in memory, let's say 00, and ends up at a certain place, and it is synchronised to the frames. There are 128 bits/line, and 224 lines: that gives you 28.7 kbits per half-frame (per field). It is synchronised so that you can separate left, right and parity. Then the block format decoder makes sure that you don't lose sync between the memory and what comes in from the video. Also you have to be very careful with video: dropouts can fool some of this circuitry, by looking just like a sync pulse. You have to have plenty of checking. Error corrections will not work if you lose sync between what's on the tape in a particular video location, and what goes into a particular RAM location. About 8K of the memory is used for data interleaving, while a further 4K is used for storing data during the vertical interval. The remaining 4K deals with wow and flutter."

The error-correction circuitry consists basically of a shift register with a number of taps, going into a parity generator. When you decode this, you compare the parity that went in with a recomputed parity, computed from the data that came in. If they don't agree, then something went wrong. "The trick is," says Adams, "to make every possible condition of bits being in error produce a unique pattern of parity errors. What we have done to simplify the circuitry is to take advantage of the fact that, on a VCR, you never really get random errors, you get bursts of errors. Using that information, you can design an error-correction system which is simpler, yet can cope with long dropout errors. We use 33% overhead: that means one parity bit for every three data bits. The parity is interleaved with the bit-stream: a bit of left, a bit of right, a bit of left, and a parity; then a bit of right, a bit of left, a bit of right, and a parity; and so on."

There is obviously going to be a variation of headroom with frequency. "It's pretty much a straight line," says Adams. "The system has over 110 dB dynamic range up to about 1 kHz. By the time you get to 10 kHz, you're down to a little over 90 dB; at 15 kHz it's about 86 dB. But fortunately it seems to follow the average peak spectrum of music pretty closely. If you have material with extreme amounts of HF you may not be able to use the full 110 dB dynamic range."

The dbx 700

The machine itself is a rack-mounting unit, 3U high, with a power supply module on the right, with switch, and input and output modules on the left, next to a pair of LED bar meters. "The whole system is modular. The unit will be available with combinations of modules, mike input, record, and playback. There will be a replay-only model, a line-level record/play model, and one with the additional mike preamps.

"The optional mike preamp has stepped gain control for each channel, from 20 to 60 dB, line/mike source select, and switchable 48 V phantom power per channel. The line input is referenced to +4 dBm. The mike circuitry offers a less than 1 dB noise figure for any impedance from 100 Ω to 1 k Ω. There is only one coupling capacitor in the whole preamp, which uses discrete transistors and is a fully-differential push-pull 'instrumentation amp' style circuit."

The input module incorporates pre and post clip lights which allow for accurate gain setting. There are three input/setting possibilities: a pot, a screwdriver preset, and an internally-set 'ref', selected by a three-position toggle switch. "With

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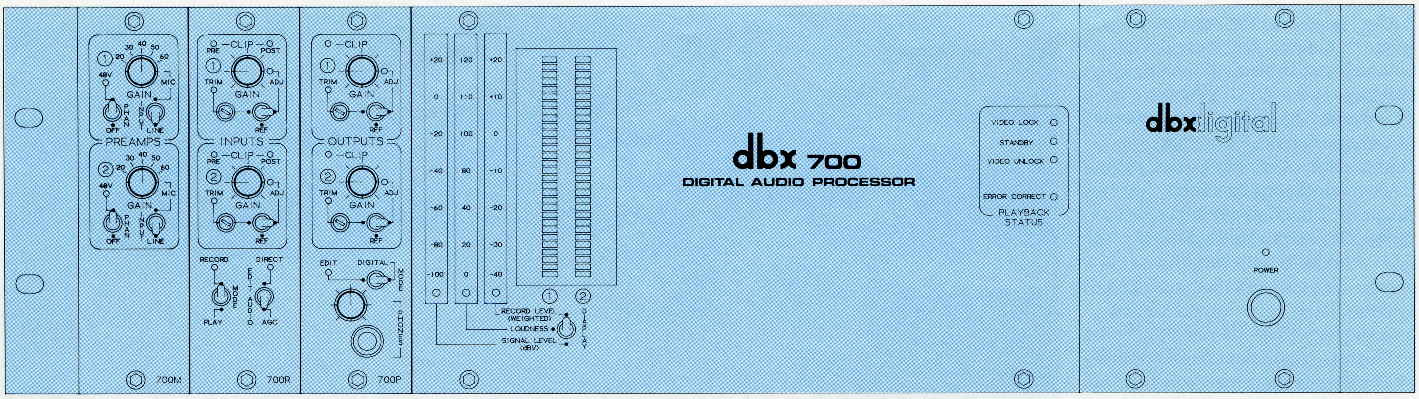

both the inputs and outputs set to 'ref', there is unity gain through the system. We at dbx feel that people should begin thinking about playing back instruments at real levels instead of the way that some studio people seem to do.'' The clip lights are either side of the gain control, the preset offering up to 20 dB attenuation, while the rotary pot goes down to infinity for fades, with 10 dB of gain available on the rotary controls. There is also a record/play switch plus switching between direct and AGC-controlled analogue 'edit' mode audio recording. ''While you are recording, there are outputs available which are designed to go to the analogue audio tracks of the VCR. These can be used for editing, where the digital audio would not be recovered in slow-motion modes. On the playback module there is a switch to select between analogue and digital, so while you are editing, you can flip into the analogue edit mode, perform the edit, then listen back digitally.'' Editing can be performed with ±1 frame accuracy with a standard video editing system. The switch selectable AGC-controlled analogue recording system utilises a 2:1 compression ratio to enable ordinary analogue VCR tracks to cope with what would otherwise be far too much dynamic range.

The output module has similar gain-control options to the input module, with a knob, preset and 'ref' position. There is also the switch for selecting digital replay or analogue 'edit' audio. The analogue ins and outs send and return to and from the VCR audio tracks, the switch selecting whether analogue or digital audio is relayed to the unit's main outputs. The module is completed by a headphone socket and level control.

The metering system has three modes. ''In the record mode, you have 2 dB per LED; 60 dB dynamic range. It is pre-emphasised to follow the decreasing headroom at high frequencies. It is very fast: 2 ms attack time. The meter is reading right off the RMS detector, where there is pre-emphasis so that for high frequencies it commands lower VCA gains. The meter's pre-emphasis is therefore exactly the inverse of the headroom, so at every frequency it will tell you the exact point at which overload will occur.

''The 'signal level' mode offers 4 dB per LED and reads in dBV, with 120 dB range (+20 to —100). It reads RMS level from either the record or replay amps. Then there's the loudness meter, which is something else. One of the things that always bothers us is the way that S/N measurements are made. A-weighting is only right for one loudness level, and is wrong for every other level. So we have made a dynamically-varying filter which will follow the Stevens curves (which are the latest version of the Fletcher-Munson curves) within 2 dB over the whole 120 dB range of the meter. That has a fixed section which accounts for the parallel nature of the curves in the 2 kHz and 5 kHz region, and a varying section which changes the low frequency response according to what level you put in. It gives you an idea of what you are really hearing, rather than 'what the machine says'. We expect most people will use the record setting during recording, and the loudness setting on replay. It's very useful to have that meter to find ground loops in the studio, for example, and things which you might not hear in analogue recording.

''On the right of the unit are a set of playback status lights. There's an error-correct light which, believe it or not, indicates that an error is being corrected! Then the other three I call the 'ready, set, go' lights,'' says Adams. ''In playback, when you have no video coming in, the bottom light is on, telling you that the video is not locked. When video comes in, and the PLL is still locking, it's in the 'standby' condition, for maybe one second, then it goes into the green, indicating that everything is go: video lock.

''Most of the components are off-the-shelf, with the exception of the VCA, which is a computer-selected 'best' version of the dbx *2151* 8-pin SIP VCA. There's a bin, containing chips that nobody will ever be able to buy from us, which go into this.''

The future

The NTSC version of the dbx *700* Digital Audio Processor will be available in the late spring, in both the US and Europe. The US price will be under $5,000: in England it is hoped to supply the unit for around £3,500 to £4,000, but this price is subject to revision, and should not be taken too literally. Shortly after the release of the *700*, a delay unit will be introduced, which will take the bit-stream and decode it after a delay, for cutting applications. There is a socket on the rear of the *700* which will supply the bit-stream to the delay unit.

One question that many will ask, no doubt, is 'what about editing?'. It is obviously possible to edit with frame accuracy on a video editor. But it is not intended to release an editing unit for the system at present. ''With the price of this product being what it is, the cost of providing an editor to do that really tightly would throw out the entire concept of what we're trying to do. Razor-blade and tight editing will be left to the next product in the *700* series, that being some form of a fixed-head, reel-to-reel multitrack machine. You don't even need to run crossfading algorithms to handle razor-blade editing, so it will be easier to implement than on PCM. Our goal is to produce a price for a machine that is very close to an analogue machine of similar track format.'' The machine should be a great success when it ultimately arrives.

It is worth noting that although the dbx system will be ideal for modern methods of conventional disc-cutting, metal mastering and so on, it cannot be used *directly* for the production of *Compact Discs*. Says Adams: ''There is no way, digitally, to convert from CPDM directly to 16-bit format. You have to go back to analogue. I don't feel that is any major drawback because I think our system sounds better than 16-bit. I don't think you'll hear any degradation going through analogue.

''There's one point that has come out of the show, which we raised at the launch of the *700*: and that's the comment that 'finally there's a digital system which *sounds* good'. We at dbx feel that 'musicality' is important. We feel that we're doing 'music' and not 'data'. We had a prototype a year before the show, and we were continually going round the loop of recording something, *listening* to it, and recording again. Because of the way this thing works, with such a high sampling and reconstruction rate, there is a feeling that because there are smaller gaps between the samples, it's probably better. People are saying that it '*sounds* good', and doesn't 'sound digital'.'' Comments like that obviously bode well for the system, and a philosophy based on 'musicality' can't be a bad thing. ■

MANUFACTURER'S SPECIFICATIONS

Channels: two.
Storage medium: video tape.
Frequency response: 10 Hz to 20 kHz ±½ dB.
Dynamic range (unweighted, maximum RMS signal to noise floor, input shorted, noise bandwidth 20 kHz): >110 dB.
Wow/flutter: less than 0.01% unweighted; less than 0.006% W RMS.
THD: less than 0.03% total harmonic distortion, 1 kHz, 1 V RMS input.
Sampling rate: 700 k bit/s.
Error correction: will completely correct 1024-bit burst error up to eight times per video frame (1/30 s).
A/D conversion: precision-companded, linear-predictive delta modulation.
Metering: two columns of 30 LEDs, switchable amongst: record level, pre-emphasized, 60 dB range; wide-range signal-level meter, unweighted RMS, dBV-reading, 120 dB range; loudness meter, 120 dB range, matches Stevens curves to within 2 dB.
Mike preamp: adds less than 1 dB to mike noise for all microphone impedances between 100 Ω and 1 kΩ; balanced in *XLR*.
Headphone jack: yes.
Console connections: line in, balanced, 10 kΩ; line out, electronically balanced; will drive 600 Ω to +24 dBm.
VCR connections: video in and out, NTSC, 75 Ω, 1 V peak-to-peak, BNC connectors; audio in (for editing), balanced, 10 kΩ input impedance; audio out (also for editing), unbalanced, drives 2 kΩ or greater.
Dimensions: (whd) 19 × 5¼ × 11½ in.
Weight: approx 20 lb.

dbx digital

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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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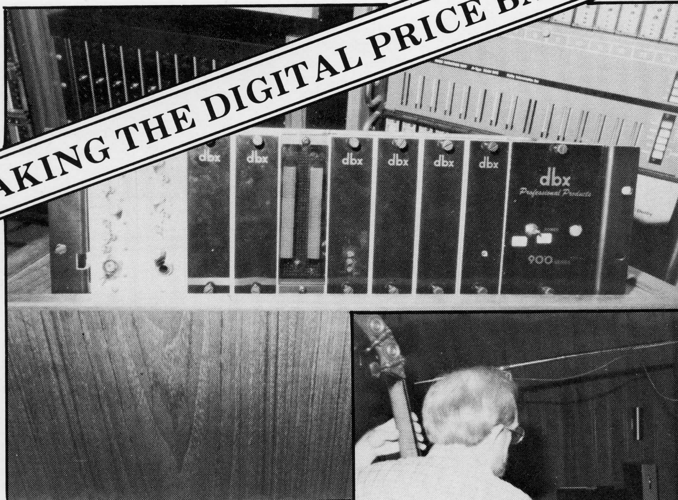
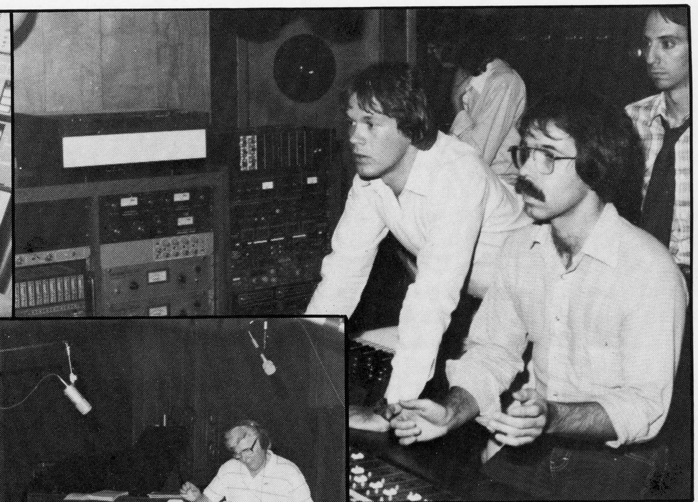
BREAKING THE DIGITAL PRICE BARRIER



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

. . .continued on page 3 —

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-

. . .continued on page 5 —

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:

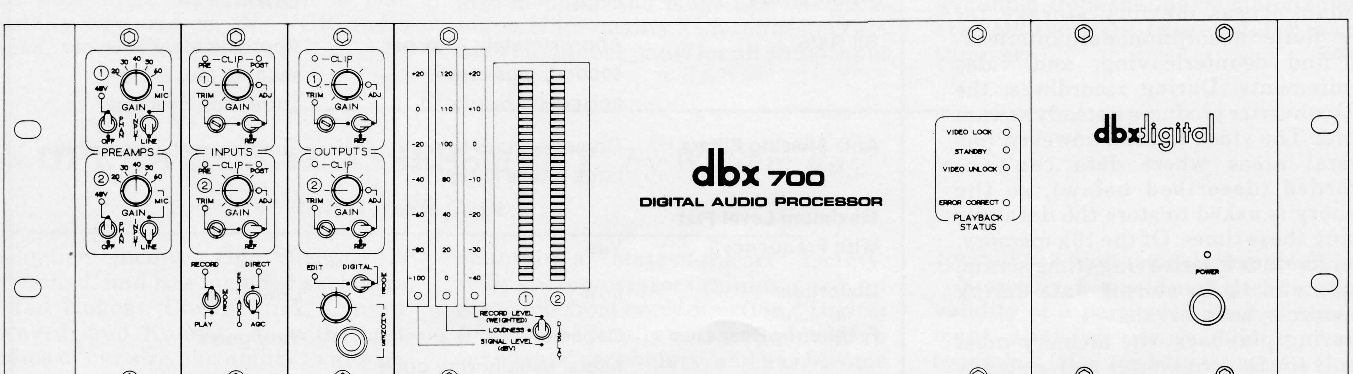
- *Companding versus Adaptive.* In ADM, step size is varied according to the average slew rate (speed of change of the input signal). A burst of high-frequency, high-level input signal requires a large step size, so that slew-rate limiting can be avoided. The problem with doing this, however, is that 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.

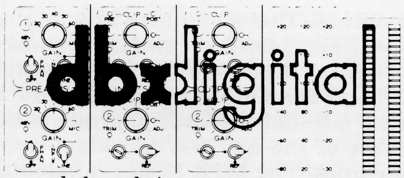
Analog Display and Control Functions

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.

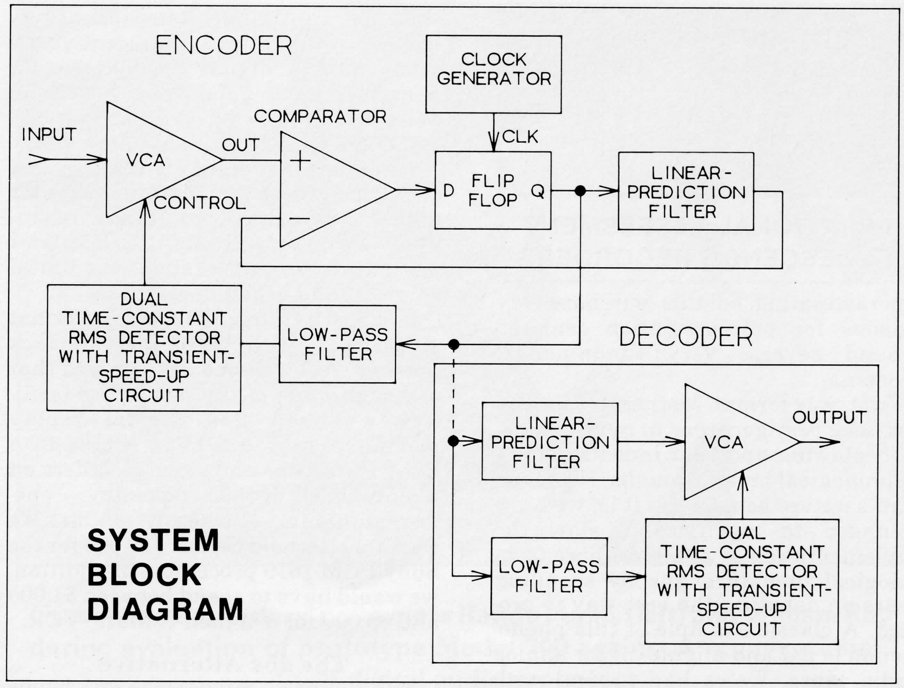
b) *Signal-Level Meter.* This is an RMS-responding, non-weighted indicator with a total range of 120 dB (+20 dBV to –100 dBV). In record mode, this meter reads the line input or output of the mike pre-amp, if one is used. In play mode, it reads the unit's line output level.

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 backplane. The complete power supply, including transformer and AC input, is also modular and plugs into the backplane. 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*

— continued from page 2 . . .

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.

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.

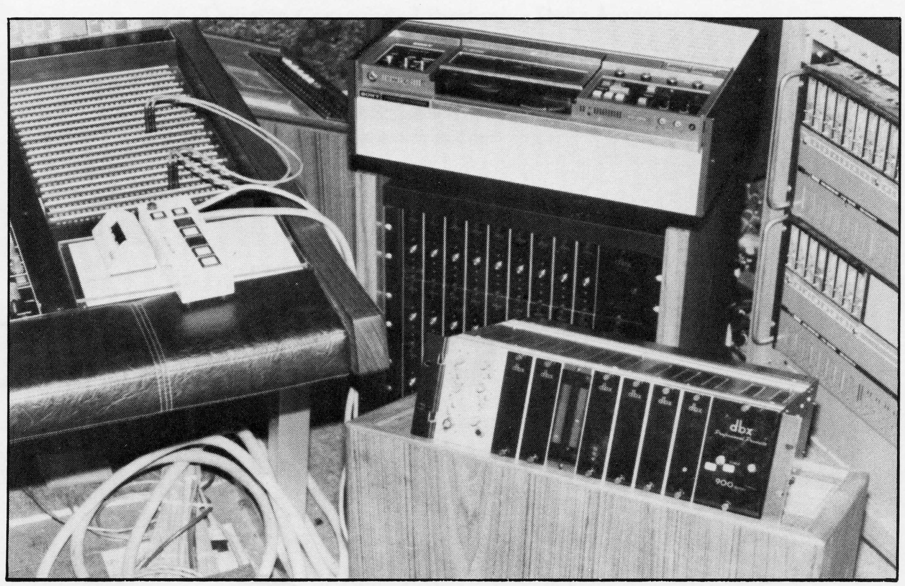
The dbx 700 uses a VCR and videotape. For editing, this means you either need two units and a video editor, or access to a video editor. However, on a positive note, any video editor that will interface with your VCR will suffice. While dbx recommends that it be used with a ¾-inch U-Matic, the 700 processor 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.

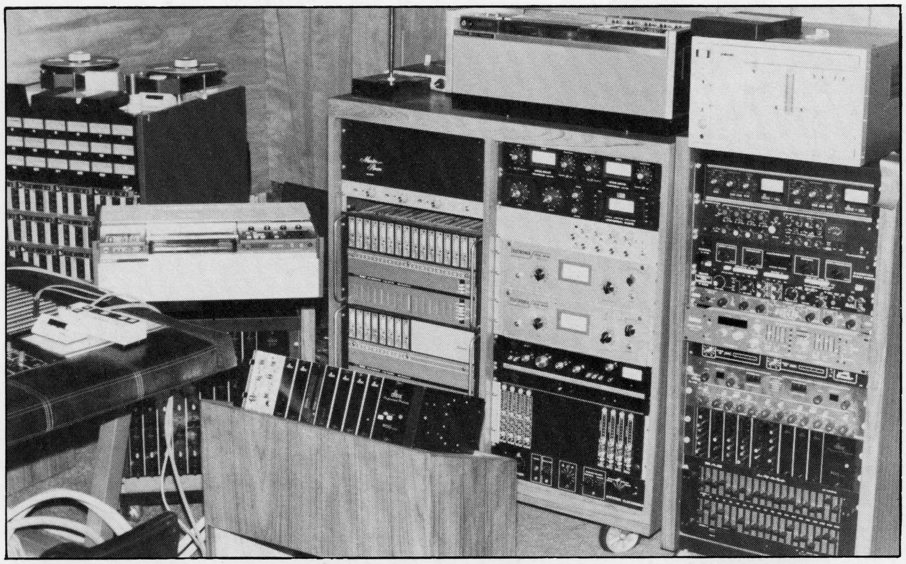
Other users of high-quality half-tracks may be interested for other reasons. One very important issue that lies on the positive side of videocassettes is that as a storage medium they are very compact and easy to handle, and you don't have to worry about record/replay EQ, or tape speed.

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

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



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. ■■■