RECORDING
ENGINEER/PRODUCER

PRODUCING AUDIO FOR ● TAPE ● RECORDS ● FILM ● LIVE PERFORMANCE ● VIDEO & BROADCAST

SUPERSTAR

by QUAD EIGHT

Advanced technology and unparalleled flexibility come together in the SUPERSTAR music recording console. Development of this console centered around the dual requirement of truly high definition sound and low noise, so critical for digital recording.

No other single console offers the combination of superior sound and flexibility in size and layout at such an affordable price. Field expandable, the SUPERSTAR provides ergonomical positioning of the console modules, allowing you to satisfy your own configuration needs. High resolution meters, central bus assignment, Intelligent Digital Faders, and the most comprehensive automation system all add up to SUPERSTAR–your next console.

MODULAR CONSOLE

The SUPERSTAR is a totally modular console using air frame design concepts for strength and rigidity. Individual frame sections are in groups of 8 modules, with plug in wiring for true field expandability. The modular overbridge accepts the new limiter/compressor/gate for use either in-line with the input module or as a peripheral.

60-segment LED bargraph meters use advanced circuitry for precise and stable indication, offering VU, Peak, VCA level, and Spectrum Analyzer displays switch selectable.

Plug-in interchangeable equalizers and preamplifiers in each I/O module give instant user selectability and allow the addition of new technology at any time. Each module is of dual-purpose in-line design with line trim, equalizer, filter, 8 echo/cue sends, and fader switchable into the monitor/mixdown or main channel. Monitor/mixdown can be assigned to two independent stereo output busses for added versatility.

CENTRAL ASSIGNMENT

This electronic output assignment cross-point switching system assures fast and reliable connections from the console to your tape machines with full routing or mixing capability. 64 output busses are assigned from each input module by a central touch control plasma display panel controlling up to a 96 by 64 electronic switching matrix. Completely software driven, the panel allows instant selection and display of the bus assignment with 10 presets in local memory. Optional unlimited storage to disk is provided. Easy to use, the system prompts for bus assignments and provides help through informative menu displays.

The building block matrix system consists of 16 by 16 switching cards bussed to 16-output summing cards. Logic controlled monolithic switching elements use zero volt current switching for extremely low distortion and feed through.

THE NEXT GENERATION

The introduction of the SUPERSTAR signals a new era in professional sound control. With more and more studio facilities acquiring digital multitrack recording capabilities up to 64-track, larger sophisticated console systems with transparent sound performance are necessary. Digital signal processing (DSP) is neither economically feasible nor technologically advantageous today. A new generation analog console with advanced digital control is required to bridge the gap between the DSP consoles of the 1990s and the currently marketed analog consoles of the 1970s. The SUPERSTAR is such a console system. See it before you decide.

AMEK announces the "soft" solution.

AMEK

Introducing the APC1000™. . .

Over twenty years ago, the concept of multi-track recording for stereo went from two to three tracks. Today, mixing 64 tracks is not unusual. The nature of audio, with digital sources and programmable electronic instruments, and the wide variety of functions demanded in recording have radically changed the requirements for an audio console.

Clearly, the answer to today's needs for audio control does not lay in yesterday's thinking. AMEK, long the leader in audio console innovation, has just re-defined audio control — the virtual console.

By removing the constraints of twenty-year-old console philosophy, AMEK has designed the obvious solution for recording, sweetening, TV Stereo, and film. The need for operational features and instant repeatability has grown but has yielded consoles of out-sized proportions and excessive electro-mechanical complexity. In-line or split monitors, and hard-wired systems, limit the operator's capabilities.

The AMEK APC1000 Large Architecture Console™ System is the "soft console" we've all waited for. Centralizing all redundant switch and control functions into one simple logic driven key entry panel has greatly expanded the flexibility, while reducing the overall size of the console and dramatically increasing accessibility. Dynamic Reset,™ Synchronous Reset™ and simplified Recall, up to 128 inputs and 64 busses are just a few of the features of an APC1000.

And, in keeping with the APC concept, AMEK has introduced the BCII, a cost-effective open architecture console system for all broadcast or production applications, featuring audio-follows-video capabilities. The AMEK BCII is customizable for your operation, yet is totally user reconfigurable.

Take a closer look at the APC1000 and the BCII . . . the soft solutions to today's serious engineering needs.

— April 1986 Contents —
Volume 17 — Number 2

News Letters Views

TISSUE PAPER PHENOMENON

from: **Bob Clearmountain**
New York City

With reference to Bob Hodas' article, "Examining the Tissue Paper Phenomenon," published in the February 1986 issue of *R-e/p*, and because I'm a close personal friend of the East Coast engineer with the "String of Hits" that was mentioned as being the originator of this technique, I would like to make a few comments on his behalf.

I believe he was quite pleased (and a bit amused) that someone actually took his "crude" and unorthodox method of high-frequency attenuation so seriously as to do such an extensive and scientific report on the "phenomenon." He was particularly pleased that some of the results (Figures 4 through 6) seemed to not only document quite closely the effect he was trying to achieve, but also confirm the results.

He was, however, disappointed that the tissue originally used — Ft. Howard Shur-Wipers (also known as "Kim-Wipes" or "Kwik-Wipes," and often found in studio control rooms) — was not tested. The results might have been slightly different in that they have a stiffer consistency than facial tissue.

He feels that the "fallacy" Bob Hodas spoke of might not lie with the "Tissue Paper" technique, but possibly with the conclusions arrived at from the final few tests. Bob Hodas concluded, from the result of covering the *microphone* with tissue, that the tissue wasn't absorbing any high frequencies at all, because the curve then remained the same. Could it be that *all* frequencies were attenuated equally, therefore resulting in no deviation on the test curve? The circumstances would certainly be quite different when *only* the tweeter of a two-way loudspeaker system is covered.

The other test the author used was to move the tissue away from the tweeter an inch or two, which resulted in the curve changing quite a bit. Bob Hodas believes that the high frequencies are merely being reflected back into the tweeter, resulting in a comb-filter effect. Could this actually be caused by certain stray, unfiltered high frequencies bouncing off whatever was being used to simulate the console surface, thus avoiding the tissue altogether? Of course, if there wasn't anything being used to simulate the surface of a console, the whole test couldn't really be valid anyway, could it?

I suppose the main point to consider here is this: even though Bob Hodas may be totally correct with his comb-filter theory, the "Tissue Paper Phenomenon" technique has been used quite successfully for some time now so ... who cares? □□□

Bob Hodas replies:

I would like to answer Bob Clearmountain's letter in two parts, first the subjective, and then the technical:

1. My article was written not so much as an attack on the origin of this "tissue-paper" phenomenon, but more as an indictment of what followed. The recording industry, in its blind headlong rush towards "hitsville," not only embraced your friend's method without question or anaylsis, but went even further by adopting its own, unproven materials. This may be evident in countless magazine advertisements and articles in which photos are shown, and engineers speak of their monitoring method. As Devo once said: "Are we not sheep" (*sic*)? Thus the "wrong type" story prompted the tissue exploration, and so there were no tests of the original Shur-Wipers. (Although I suspect from my past data that similar results would be found with Shur-Wipers, conclusions cannot be drawn without proper measurements.)

2. Your assumption that tissue will attenuate *all* frequencies equally is purely and simply *incorrect*; I feel perfectly safe in saying that any acoustician would back me up on this fact. However, even if this were the case, the resultant graph would

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From the Editor:

One characteristic of the Pro-Audio Industry that impresses me greatly is its ability to take existing technology, and adapt it to meet specific needs. Let's not kid ourselves. Apart from one or two notable exceptions, our marketplace is simply too small for manufacturers to develop most of the components necessary to put together leading-edge audio production systems. Having to overcome the inevitable "economy of scale" factor precludes mass production of many of the key circuit and system building blocks we need to develop, for example, digital tape machines, effects processors, synchronizers and other hardware to make us more creative in the studio and on the road.
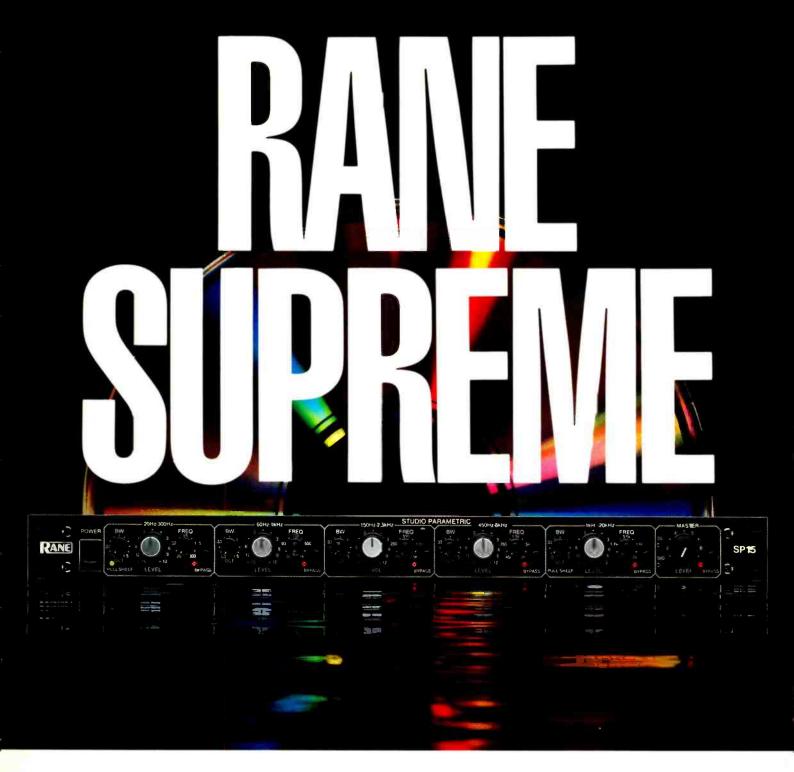
Yet there already exists a virtual treasure trove of microprocessors and support chips, not to mention proprietary operating systems with which to program them. Now that complete micro- and minicomputer systems are available off-the-shelf for a fraction of the cost of last year's mainframes, we are beginning to see the emergence of a new generation of software developers that are able to come up with very user-friendly packages for synchronizer control, console automation, MIDI sequencing, session bookings ... you name it, and there's a very good chance that somebody is writing the software and/or modifying existing hardware to do the job for you.

A prime example exists in the current generation of random-access editing systems being developed by several companies and facilities around the world. Obviously, now that digital recording and processing are becoming *de rigueur* in a growing number of production facilities, we will see more and more studios and manufacturers looking at ways of manipulating digital data streams, in much the same way as we now take word-processing and spread-sheet capabilities for granted. Although I would readily acknowledge that the development of even a basic digital editing system represents a far from trivial exercise, we do at least have access to some extremely powerful — and affordable —building blocks with which to work.

If I have one word of caution, however, it is that the wave of the not too distant future will be towards system integration. This year will see the release of several high-speed parallel processors that out-perform our current micros in terms of computation speed. Given the quantum leap in processing power that such devices represent, what I hope to see is a move towards integrating the type of discrete functions now provided by separate systems, and thus provide centralized control of production tasks. It seems to make little sense, in terms of both operational flexibilty and hardware costs, to provide ourselves with separate, dedicated systems for, let's say, console automation and timecode synchronization, when these functions are complimentary and could easily be handled by one system. The new designs of virtual or assignable consoles — utilizing either all-digital processing, or digital control of analog functions — will allow a greater degree of integrated functionality than today's configurations. While centralized control of outboard equipment connections will remain a viable alternative for many years to come, I predict that we are not too far away from the day when practically every audio recording and processing function comes under the control of a master computer that can be reconfigured to handle virtually any task we can think of. **MEL LAMBERT**

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have shown a trace very much like Figure 3 (upper), only it would have appeared lower on the chart, indicating attenuation by several decibels. But this was not the case, and the trace showed no signs of attenuation. (That particular chart was not published for reasons of space — it is, after all, essentially the same as the reference line — but I do have it on file if anyone wishes to examine it.)

As to his reference to the measurements taken at various distances, results in both a console environment and a free-field environment demonstrate the same characteristics of shifting frequency dips, which indicate a delay-type cancellation. The point to be made here is that frequency response is more dependent on *positioning* of the tissue, rather than tissue *type*. The use of a tissue-paper layer is basically a non-repeatable, uncontrolled method of filtering that will change from studio to studio.

To finalize, I can only hope that one would desire the control that a filter provides, as opposed to the inconsistencies that the tissue layer demonstrates. It's no wonder that it has taken so long for Sound to be taken seriously by such sophisticated industries as film and television, when we operate with methods that could fit into the script of *Spinal Tap*! I firmly believe that as we move ahead into the digital world, we are going to have to demonstrate a more knowledgeable and critical approach to our recording art. ■■■

MICROPHONE EVALUATIONS

from: **Klaus Keyne,
San Francisco, CA**

As importer, collector, and restorer of innumerable Neumann tube microphones, I approached Professor Lowell Cross' evaluations [December 1985 issue] with considerable apprehension. But, after reading the article, I was delighted to find no bias on his part for either tube or transistor designs in general, or brand preferences in particular. Despite repeatedly professing subjectivity in his deductions, Professor Cross is actually right on the money with his personal evaluation of the respective makes and models, reflecting pretty much the general consensus of hundreds of experienced engineers and studio-owners I have talked to over the years. *R-e/p* should make reprints available; this was an excellent overview and primer for novices in the field of studio condensers!

I would like to supplement December's article with two random observations, on tube mikes in particular, condensed from a decade of experience and experiments:

1. Regardless of brand, there can be a tremendous variation in performance characteristics between any two seemingly identical, healthy looking microphone capsules of the same model, the same series, even consecutive production numbers, since hand-manufacturing, -assembly, and -diaphragm tensioning; aging, climate influences; use and abuse can be responsible factors.

Mike amps do not fare much better. Tol-

erances in parts like tubes, resistors, and transformer wire; aging of capacitors; changed impedences through tube wear; and invisible dirt on multi-megohm resistors add up to a noticeably different sound in identically appearing amplifiers.

Considering the cumulative effect of these factors, I am always amused when clients request a "matched pair with consecutive serial numbers" for use as a stereo pair — instead of selecting similar sounding capsules for similar sounding amps.

2. Microphone design requirements by broadcasters in the Fifties and Sixties dictated different parameters for gain, frequency response, etc., than are desirable and possible today. Transient response, self-noise, and frequency range of tube-mike amplifiers can be noticeably improved through minor surgery, thus bringing out every nuance a particular capsule has to offer, without altering the overall "personality" of the mike in the least. The evolution in tube circuitry, and new materials like polypropylene for capacitors, also allow subtle improvements, analogous to a pop-filter being removed from the microphone.

Recent impressive advances in transistor-mike design confirm my gut feeling: the tube condenser is still the standard bearer. But these second-generation transistor challengers can only be kept at bay by diligent maintenance and fine-tuning of the old magic! ■■■

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"Letters to the Editor"

... *continues on page 12* —

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SONY APR-500 REVIEW: REPLY FROM THE AUTHOR

from: Peter Butt
San Pedro, CA

Regarding the Letter to the Editor from Stanley P. Lipshitz and John Vanderkooy, published in the February issue of *R-e/p*, all of the points raised in item #1, concerning the properties of a symmetrical bipolar squarewave, are commonly and generally recognized. The desirability of a dual-channel data acquisition device also is recognized and favored, but such is not at hand. The long-recognized limitations of the spectral properties of the squarewave are dealt with by empirically choosing a tape-drive level for the analysis signals, such that a usable compromise between tape saturation, noise floor, and 12-bit digitizer dynamic range can be achieved.

I take care to do this, being guided by empirical observation of indications of signal-waveform compression prior to undertaking a data sample. The transfer functions presented in all of my reviews are the result of the "splicing" of several deconvoluted data sample waveforms of different fundamental frequencies. Each time-domain data point is taken as the arithmetic mean of at least four data samples. A set of sample data is discarded, adjustments made, and re-taken if the deconvoluted magnitude and phase data do not form reasonably continuous curves that do not agree with known typical signal-channel performance. For the case of parameters such as complex impedence magnitude, absolute gain and phase, discrete points are routinely taken by alternate method as a check on the accuracy the results of digitized data.

As a further check on the empirical phase response obtained in the case of magnetic recorders, it is often possible to detect the polarity (not phase) reversal occuring at the wavelength of reproducer-gap magnitude null. Effective reproducer gap-length determination has been possible in the case of at least one of my reviews published in this magazine. In the case in point, the magnitude nulls in the Sony/MCI APR-5002 7.5 ips Record/Sync Reproduce response are clearly evident and pointed out in the original review. It can be seen that the effective record-head gap length is about 500 microinches. Examination of the relative phase data shows the polarity (not phase) reversal for that case.

I admit, and have long recognized, that the choice of the bi-polar squarewave and the single-channel time-domain sampling method represents a compromise, but is one dictated by existing facilities and time limitations. Carefully used with recognition of its limitations, the technique can and, I believe the evidence shows, does yield representative characterizations of the equipment evaluated. More on this point below. The bipolar squarewave is like any other tool: its utility ultimately rests with the user.

As far as the validity of a bipolar squarewave to the task of characterizing an audio processing/storage device is concerned, I can only suggest that observation of the transients found in the production of acoustically-generated music might be instructive. Percussive sounds, such as cymbals, tambourine, claves, metallic keys, etc., are easily capable of overloading *any* analog magnetic-recording system, even with careful level management. Indeed, many microphone pre-amplifiers are easily overloaded by such common acoustic stimuli, and typically exhibit squarewave risetimes of about 5 microseconds, or more. In fact, I have come to believe that it may not be currently possible to even approximate accurate reproduction of a harpsicord or even the common (?) concert grand piano through an electric channel having electro-acoustic

Figure 1
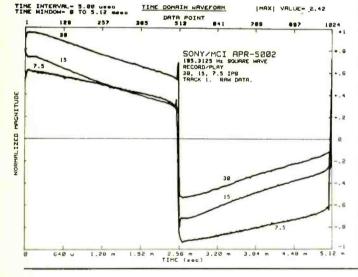
Figure 2
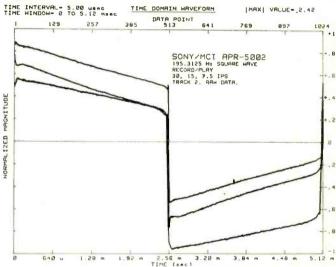
Figure 3
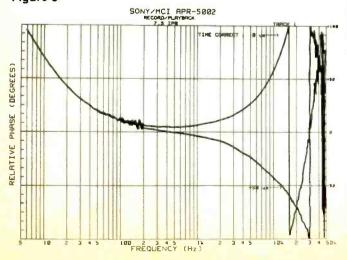
Figure 4

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transducers at its input and output, without audible compromise in linearity and/or dynamic range. This statement leaves magnetic recording and reproduction of those instruments completely aside. The acoustic experience of these, and probably other, such musical instruments is *never* accurately reproduced electronically in my limited experienced. The band-limiting in *any* recorder signal channel is more than adequate to prevent the undistorted transmission of naturally occuring transient wavefronts having rise times on the order of about 2 microseconds, perhaps less. Practical reproducer head-gap lengths (200 microinches) preclude the reproduction of rise times shorter than about 7 microseconds. This is so even at 30 ips tape speeds.

Hearing the replay of a sparkling acoustic performance, just rendered, can be a frustrating and disappointing experience. I invite the gentlemen from the University of Waterloo to partake of it some time.

The name of the game here is to evaluate the ability of a given professional audio processing device to accurately pass an acoustically-generated signal. Toward crude definition of some properties of such a signal, I have observed commonly encountered acoustic sound pressure level spectral distribution envelopes using third-octave devices. The results of these casual surveys has shown that most close-miked sound fields generated in the course of ordinary dialog and other activities in office environments and in dramatic stage situations, again using close-proximity microphone placement, possess magnitude envelopes that are approximately flat-spectrum to about 10 kHz, falling at about 20 dB per octave above that frequency band. Others may make this observation at their leisure. The spectral distribution of contemporary musical program material is generally observed and intended to approximate a flat-spectrum energy distribution envelope, and is commonly recognized so as a matter of course. Again, these empirical observations are available to all who care to examine daily acoustical experience.
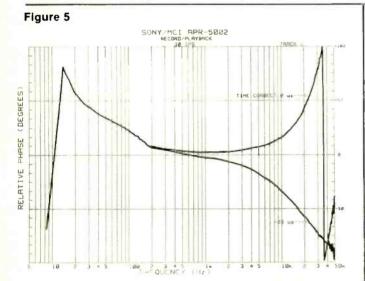
I am inclined to believe that the convenient choice of squarewave signals used in my evaluations is a rather charitable one, in view of the 6 dB per octave decline in harmonic level of that signal spectrum with increasing frequency, compounded by their low fundamental frequency. If system channel overloading does occur within range of the common squarewave, that is the property of the device that I wish to expose and examine.
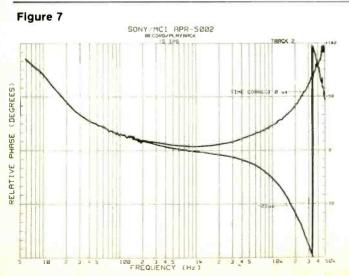
As far as the gentlemens' point #1 is concerned, I too would wish to be able to employ yet more sophistication in measurement technique, and data analysis and presentation. I readily agree that improvements can be made. Indeed, are there any areas of human endeavor where the same criticism cannot be levied? I require only the discretionary after-tax windfall of about $70,000 to make all of us much happier.
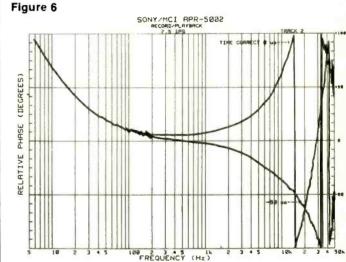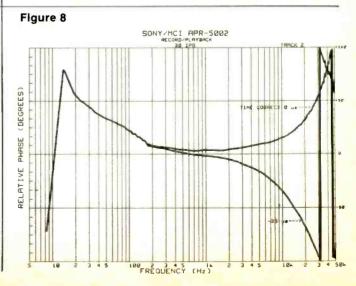
I give no quarter here. I must *emphatically* disagree that my methods are carelessly, inaccurately, ignorantly or fraudulently applied. I am surprised that, informed and knowledgeable as I know the gentlemen to be, their readings of my work, and their impression of me from the several brief interractions we have shared, would inspire condescending questions implying my ignorance and/or disregard of such vital elementary matters bearing so directly on the validity of work that I prepare for public view.

Turning to item #2 in their letter, a single-channel digitizer is indeed the means for my digital-data acquisition —it is the device of necessary choice. I am aware of the effects of flutter and their generally inverse relationship(s) to tape speed. I have found situations where the magnitude of peak flutter observed in a tape machine precluded characterization of the record/reproduce transfer function

Figure 5

Figure 6

Figure 7

Figure 8

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through any significant portion of the usable (?) bandwidth of the device using digital evaluation techniques. In cases where this trouble has been encountered, the device was either withdrawn by the manufacturer, or necessary repairs made and the evaluation repeated. In *no* case have I ever presented data that I had reason to believe to be inaccurate. In cases where I have been uncertain of anomalies in data or conflicts in interpretation, I have always chosen to isolate and correct the cause of those inconsistencies or, failing that, candidly reported them in my text. My record stands in this regard.

Once again, I wish to declare to Messrs. Lipshitz and Vanderkooy, and to all others, that I have had extended intimacies with the behavior of analog magnetic-tape transports used in audio, video and instrumentation applications. I recognize the impairment of non-linearities and uncertainties resulting from system noise, quantizing error, and non-synchronicities in digital sampling and subsequent Fourier transformation data manipulation and analysis.

By way of explanation, the digitizer in question, a Hewlett-Packard 3437A System Voltmeter, is generally triggered from the sync output of the squarewave generator — this is done in cases where propaga-

tion time delay in the device under test is fixed, which is the case for an amplifier of some kind. For the case of a tape recorder/-reproducer, the time delay between input and reproducer output is variable from the order of milliseconds to years. For reasons now known to us all, it is impossible to compensate for the variations in time delay between the record input signal and the reproducer output to provide accurate time reference for sequential evaluation of the reproduced signal. Add to this the mechanical elasticity of the tapes commonly available, and typical of popular recording stock, and it is seen that even the fairly short delay due to the transit time of the tape between record and reproducer head gaps is highly variable beyond compensation. I have recognized this problem early on. Triggering the digitizer from the generator for the case of record/sync response is obviously impossible, because of the mutual exclusivity of those simultaneous record and reproduce functions for the same magnetic head.
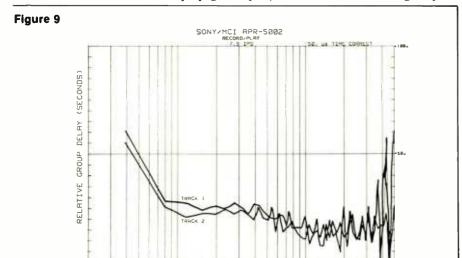
The technique I have used to generate the trigger pulse necessary to synchronize time-domain data acquisition is to take advantage of the adjustable voltage level and selective slope triggering capability of an oscilloscope. In this way the data acquisition cycle can be reliably begun at the same point in the reproduced signal cycle, if care is taken in selecting the point

at which the oscilloscope sweep is set to unambiguously trigger. This convenient point often turns out to be near the maximum peak value of the reproduced signal. In choosing to take this approach, the tenuous absolute relationship in time between the driving signal and the reproduced signal is necessarily lost — if such a relationship may exist at all, that is. The alternative to this approach is to forego the attempt to determine the record/reproduce transfer function, and accept only the continuous-wave sweep data reported by, I dare say, all other equipment reviewers.
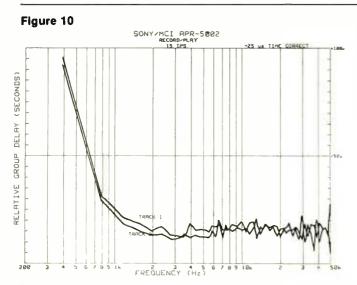
A side benefit garnered in exchange for the sacrifice of absolute-time coherence, is the shortened time of one, single-signal period per sample for flutter degradation to be operative. For a 512-point time-domain sample, peak flutter would need to be about $[1/(512 \times 2)] \times 100$ per cent peak, or about ±0.1% peak to shift the synchronicity of the sampling interval by one data point. This level of flutter performance can be achieved by most tape machines claiming the title of "professional." If not, I candidly hold that the device is not worthy of review.
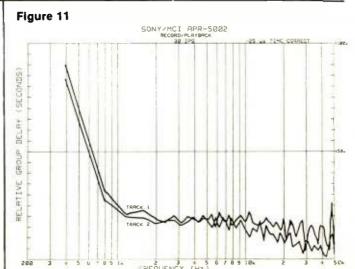
Another approach is to choose a data sample having a greater number of time-domain data points. The time jitter will degrade the frequency domain data at the high-end of the spectrum to a much greater degree than at the lower. The useful, valid data at the low end of the frequency domain may be kept after verification, while the noise-dominated higher spectrum data is then disregarded. This is a legitimate empirical analysis technique, as the gentlemen are well aware.

A factor not touched upon by the gentlemen is the inherent variability in the magnitude of magnetic-oxide recording sensitivity of the tape stock used in making the test recording, which is attributable to variations in raw stock, chemistry, and manufacturing processes. Variations in medium-wavelength recording sensitivities of ±0.25 dB are common over the head-to-tail of 2,500-foot lengths of tape. In some cases, noticeable variations in reproducer output, specifically *not* attributable to tape skew, as great as ±0.5 dB peak-to-peak, sometimes greater, can be observed over time periods as short as two or three seconds. Tape manufacturers cover themselves by specifying head-to-tail sensitiv-

Figure 9

SONY/MCI APR-5002
RECORD/PLAY
7.5 IPS   50 uS TIME CORRECT

TRACK 1
TRACK 2

RELATIVE GROUP DELAY (SECONDS)
FREQUENCY (Hz)

Figure 10

SONY/MCI APR-5002
RECORD/PLAY
15 IPS   25 uS TIME CORRECT

TRACK 1
TRACK 2

RELATIVE GROUP DELAY (SECONDS)
FREQUENCY (Hz)

Figure 11

SONY/MCI APR-5002
RECORD/PLAYBACK
30 IPS   25 uS TIME CORRECT

TRACK 1
TRACK 2

RELATIVE GROUP DELAY (SECONDS)
FREQUENCY (Hz)

For additional information circle #9

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ity variations as great as ±0.5 dB within a given reel, and as much as ±1 dB between reels, My experience indicates that these are optimistic expectations, and should not be blindly relied upon. I take care not to.

For this reason, it is not possible to use the entire length of a given reel of tape, and the choice must be made to use results of only a small portion of it. Again, this is done in my evaluations as indicated in the interests of unambiguous, accurate data.

I do not infer that I would take tape manufacturers to task for this deficiency, since I have long believed that the making of tape is an art reserved for leprechauns and other spiritual beings.

Continuing on, yet another perturbation that could possibly lead to errors in digital data sampling and, therefore, in determination and interpretation of results, is the troublesome tendancy for recorded magnetic fields to reproduce themselves, after a fashion, on adjacent tape layers, as they are wound onto their take-up reel. Add to this interlayer "print-through" effect the tendency of a magnetic wavefront gradient to disperse itself longitudinally along the tape, reducing the magnitude of the remnant magnetic fluxivity gradient from its initial level upon leaving the trailing, collapsing magnetic field of the recording-head gap. Both of these effects are thoroughly audible and verifyable by direct experience: i.e., *by listening to music.* The interlayer printing phenomenon is well known to all, and frequently specified as a performance parameter by magnetic-tape manufacturers who have nothing to be ashamed of. The values of interlayer print attenuation depend on recorded signal wavelength, but are often observed to vary from about -50 dB to about -65 dB below reference signal level in peak magnitude for adjacent tape layers. I have repeatedly observed these levels of interlayer printing over time periods as short as 10 to 20 seconds.

The quantifiable results of magnetic transient longitudinal dispersion are unknown to me and perhaps to most other people. I have never seen any published data regarding it.

The high-frequency roll-offs objected to and summarily condemned as "wrong" in my original review by the gentlemen are entirely a consequence of the Discrete Fourier Transformation of the APR-5002's time domain reproducer output, and its complex division by the Discrete Fourier Transform of the reference signal. I am sorry, gentlemen, but I cannot claim or accept responsibility for the behavior of the physical world. What is, in the end, is what is.

The time-domain waveforms are not unusual for any comparable machines; they are correct within their quantization resolutions and accuracies and noise floors, and their necessary consequences are entirely valid in every respect.

All of the record/reproduce magnitude responses originally given are as valid as ever; I take this opportunity to re-emphasize their accuracy.

With regard to the validity of Figures 7 thru 9 and 12 thru 14 appearing in the original review, they represent valid data

and represent the performance of the Sony/MCI APR-5002 fairly, if not flatteringly. More will be said about this in my response to the gentlemens' item #3 below.

Again, with regard to the facilities used in these evaluations, and the reasons for their choice, my response is detailed in item #1 above.

Addressing the points raised in item #3, I have spent some time examining the data presented in my original APR-5002 review. I have chosen to reproduce some of that data here as it is highly relevant to the questions at hand.

The reader and the gentlemen are requested to please direct their attention to Figures 1 and 2 reproduced here, and which show the raw time-domain record-reproduce data taken from a test of the APR-5002. These data files are the data base for *all* transfer function data so labeled in that published review. The squarewave traces are shown vertically displaced for clarity.

The APR-5002 was previously aligned for the proper overbias at 1.5-mil wavelength for each of its three speeds. Record equalization was set in the customary manner, adjusting for flat response between the commonly used reference points of 1 and 10 kHz. A brief check for flatness of magnitude response was then made throughout the band and beyond, all this using the customary low-distortion sinusoidal CW signal. Low-frequency reproducer equalization was then adjusted for 0 dB magnitude response at a low frequency judged to correspond to the worst maximum contour response undulation peak for each speed and each track.

Inspection of the two figures will reveal that the data sample point number one begins at a point slightly below the absolute positive peak of excursion. This is the point to be used as the data-sample trigger point described in my response to item #2 above, using the oscilloscope triggering technique.

It is relevant to note that the synchronization pulse available at the function generator sync port occurs in exact synchronization with the onset of the positive rise at the function generator signal-output port. (This is generally true of all meritorious function generators with which I have had experience.) The point of all this detail is that, because of the practical necessity of triggering the start of the reproducer squarewave signal near its peak value, thus losing the precise absolute-time relationship between signals at the device input and output, the reproducer data sample *appears to lead* the positive-going edge of the generator signal by some small interval of time. The importance of this, still further, is that the magnitude and phase information implicit in the digitized device output waveform is arrived at by complex division of the output data sample by the generator signal-data, and is representative of an output signal that *appears* to precede the input signal.

Inspection of the reproducer output squarewaves will reveal the characteristic appearance of delay-compensated transient response: that is, the spike occuring slightly before the onset of a transition slope. In systems that have an *absolute phase* response which is generally lagging with increasing frequency, we are accus-

tomed to seeing a spike or other kind of overshoot following a transition slope. It is this characteristic to which the gentlemen from the University of Waterloo refer when they judge that the traces in Figure 10 of the original review show "this phase compensation to be accurate." I take this statement to mean that the system group delay is flat for the frequency band above the fundamental frequency of the driving squarewave. The general rule-of-thumb in making such judgements from time-domain data is that if the pre-transition and post-transition spikes observed in the case of a squarewave appearing at a system output are of approximately equal magnitude, the system may be said to be compensated with respect to relative group delay for the frequency band above the fundamental frequency of the driving squarewave. I hold that the compensation may perhaps be optimal.

Notice, please, the difference between the squarewave traces for tracks 1 and 2, if only with respect to the presence or absence and relative magnitudes of the pre- and post-transition spikes. Remember that the respective channels were biased and equalized as closely as my eyes could determine, using an external sinewave generator having demonstrated amplitude flatness and using the VU metering presumably provided on the APR-5002 for indication purposes. Notice the differences in the waveforms for nominally indistinguishable conditions under circumstances prevalent in application. Remember, also that the APR-5002 does *not* leave the degree of delay compensation to the operator, and that the compensation is determined by the operator's choice of magnitude of record equalization only. I submit that, by the pre/post transition spike criteria, no two channel/speed combinations are seen to be "accurately" or even uniformly delay compensated. Can the reader judge the degree of delay compensation for each individual case and, further, state that the amount is "accurate" or "inaccurate?"

Is there no possibility that the relative height of the pre- and post-transition spikes might be related to the specific nature of the uncompensated signal-channel transfer function? Is there knowledge available from any quarter as to this fine point of gestalt waveform analysis? I cannot say one way or the other.

If Messrs. Lipshitz and Vanderkooy can indeed make such a judgement from the mere visual inspection of a squarewave trace, I must give them credit for being far better men than I. How obvious is their conclusion regarding the "accuracy" of the degree of delay compensation displayed by the time-domain responses shown? Let us proceed further.

The accompanying Figures 3, 4 and 5 show phase response of the APR-5002's functions for track #1 record/reproduce at 7.5, 15, and 30 ips. Figures 6, 7, and 8 show the same phase data for the transfer function of track #2. The magnitude data for each case remains identical to that shown in the appropriate figures included in the original APR-5002 review, published in the August 1985 issue.

Notice, please, gentlemen and dear reader, that each of these figures shows not one but *two* phase traces. In each case

LETTERS

the upper trace shows the *relative phase* response as indicated in the presentation that prompted the academic inquiry and summary final judgement as to the validity of that data by the gentlemen from the University of Waterloo. That data presentation was generated by adjusting the delay offset of the data set below 200 Hz to match, approximately, the unadjusted data above 200 Hz. (The frequency in question is 195.3125 Hz.) I admit that the choice was arbitrary; it was not *wrong* — it was, perhaps, unconventional. It should not have been misleading to the knowledgeable gentlemen, had they taken care to note that the vertical axes are clearly labelled as *relative* rather than *absolute phase*. In any event, the result of this arbitrary choice was to show the relative phase, not the absolute phase, response of each channel with an increasing lead rather than the decreasing lead (or lag) customarily observed for the absolute phase case.

Overlaid on each of Figures 3 thru 8 is the *same data* as appears in each of the upper traces, but this time the phase data is time-shifted by the time interval noted on each figure for each of the cases shown [either 25 or 50 microseconds]. The consequence of this is merely to shift the time reference for the time-domain data sample point number one to the left. This shift is earlier or negative in time; the time-domain trace is thus shifted to the right.

If the reader, and gentlemen, will refer to the traces of Figures 1 and 2, and refer to the explanation for the late onset of digitization, and the *apparent* early advent of the reproducer output squarewave given in item #2 above, this should be seen to be entirely reasonable as well as incontestably valid. Further, doing so is *right*. No other change was made. I submit that there was no intention to misrepresent or mislead, and certainly no knowledgeable reader should feel justified to have taken an alternative position.

It is for this very reason that the phase data graphs shown in all of my reviews, not only for the APR-5002, are carefully labelled "*Relative Phase (Degrees).*" I believe that I have been careful to label all such graphs of record/reproduce transfer functions in this way, and will continue to do so in the future as may be appropriate.

Proceeding still further, I have taken the liberty to display the *Relative Group Delay* plots for the phase traces of Figures 3 thru 8. Casual examination will show that each of the curves, though noisy, is not trendless. One may easily fair some kind of linear approximation through each of the flat portions by more or less sophisticated means.

Figure 9 shows the Relative Group Delay data for tracks 1 and 2 at 7.5 ips with two plateaus: the one above about 5 kHz is lower than the other by about 10 microseconds. I would judge these channels to be slightly over compensated, as indicated by these Relative Group Delay curves. By similar argument, I would judge the 15-ips cures of Figure 10 to show slight under compensation. Again, the 30-ips delay curves would indicate slight degree of over compensation, track #1 a bit more than track #2.

Again, I remain *right*.

To summarize:

1. The input signal may, indeed, be non-optimally chosen in the estimation of the gentlemen from the University of Waterloo. Then again, it may not. I submit that this point is a "Chocolate-versus-Vanilla" controversy. A more popular question would be: "Is the test signal chosen capable of rendering the results required and claimed?"

I submit to all that the demonstrated answers to this more relevant question is: Yes, it *is* adequate for the purpose at hand.

2. For the matter of the averaging being the cause of the "high-frequency measurement error," summarily condemned by Messrs. L. and V., I can only say that, as in the case of *any* empirical measurement, error exists in the data shown. I sin no more than those who throw the stones. This error is not excessive, nor an obstacle to the clear understanding and interpretation of the data presented. Existence of error in empirical results would disqualify any and all empirical data, by the gentlemens' implication.

I enter a plea of *no mea culpa* on this one. The criticism is always valid to some degree, but irrelevant here.

3. On the point that the "phase responses are wrongly time-delay corrected," the gentlemen make a very interesting choice of words. Notice, please, reader and gentlemen, that the words "time-delay corrected" are used in reference to the offending data presentations. They pronounce these same referenced graphs as "incorrect" and "acausal" in their Prologue item #3. I believe that all of the graphs appearing in the original APR-5002 review are candidly labelled *Relative* and *Absolute*, as applies. I remain puzzled that any such clearly knowledgeable and reputable individuals could have been perplexed and, presumably, confused by misunderstanding the significance of the descriptive terms "Relative Phase" versus "Absolute Phase" and "Relative Group Delay" versus "Absolute Group Delay."

I must say that my graphic presentations are not *wrongly* but *unconventionally* time-corrected. None of the data of the original review are in any way, excepting for the noise floors, "acausal." Nor is any of the data I have presented been shown to be "wrong" in any significant way. There is, I submit, a very great difference between data that is "wrong" or "incorrect" and data that is "unconventional" in its presentation. I admit that a bit more thought on my part would have probably led me to a different choice of time correction.

Because the group-delay function is related to the second derivitive of phase with respect to frequency, the conclusions I expressed initially regarding the degree of relative-delay compensation are, in fact, still correct. The data and my position regarding the Sony/MCI APR-5002 review remain unchanged.

I express my thanks to the gentlemen for their lengthy useful comments. I welcome the opportunity to clarify any misconceptions concerning my data or methods. I remain perplexed as to how my presentation would lead to questions such as these.

I must hold that I see no necessity for a revision. ∎∎∎

prices. Choose from 10 different metering modules, 3 multi-track monitor options (including separate monitor EQ), mono or stereo faders, audio subgroups, automation compatible VCA groups, video switcher interfaces, subgroup reassignment modules, up to 3 solo systems, multi-function test generator, input selectors, limiters, compressors, patchbays with bantam or ¼" systems, and up to 10 auxiliary busses.

Basic input modules feature 3 or 4 band EQ, microphone/line inputs, 5 pre/post-fade auxiliary sends, and channel overload indicators. Options include transformerless mic pre-amps on a subcard, separate transformerless TAPE input for remix, stereo input modules, stereo EQ, internal stereo X-Y/MS active matrix, stereo blend control, dual line inputs, variable HP and LP filters, user defined panel switches, and the list goes on.

Listen to the quiet. The entire 900 console frame design is consistent with the advanced module design. A completely independent signal reference ground system assures preservation of individual circuit CMRR figures. The result is overall noise performance compatible with digital recording.

As time goes by. All 900 consoles adhere to strict Studer standards for precision and reliability. The frame is built on a rigid channel and brace structure, and each module uses pin-and-socket Eurocard connectors. Frame connectors are mounted on longitudinal master boards with solid support from horizontal and vertical frame members. All components, switches and pots are commercial/industrial grade from the best U.S. and European manufacturers. In sum, a 900 is built to last as long as a Studer recorder.

The Swiss alternative. If you have been considering a high quality mixing console from any American or English manufacturer, you should also look closely at the Swiss-made Studer 900. For quality, flexibility, and reliability, it ranks among the world's finest. Also, you may find the pricing surprisingly competitive.

For more information on Studer consoles, call or write: Studer Revox America, Inc., 1425 Elm Hill Pike, Nashville, TN 37210; (615) 254-5651.

STUDER REVOX

AES STEREO AUDIO FOR TV AND VIDEO CONFERENCE

The fourth Audio Engineering Society International Conference on Stereo Audio Technology for Television and Video, to be held May 15 thru 18, Rosemont, IL, will address the "new problems and new opportunities for the audio industry resulting from the intense demand for this emerging technology."

According to conference organizers, experience to date has demonstrated that "conventional audio practices in broadcast recording are inadequate to meet the needs of stereo audio production. Problems of phase, level match, lack of standards, the need for a solid center-channel image, and economic uncertainties require attention and practical solutions."

To promote industry growth through better understanding of stereo audio for television and video, conference chairman and AES presdient Robert B. Schulein has organized a three-day program of technical sessions, demonstrations, and panel discussions. The conference will be held at the Westin O'Hare Hotel, adjacent to O'Hare International Airport in the Rosemont area of Chicago. Registration costs will be: AES members, $345; nonmembers $395. Registration includes conference preprints.

Session chairmen include Don McCrosky, ABC Television; David Robinson, Dolby Laboratories; Bill Varney, Universal Studios; Emil Torick, CBS Technology Center; and Martin Polon, Polon Research International.

For further information, contact AES, 60 East 42nd Street, New York, NY 10165-0075. (212) 661-8528.

RUPERT NEVE ADDS KEY PERSONNEL TO WEST COAST OFFICE

According to company president Barry Roche, "Neve is expanding the West coast office to meet the rising demand for Neve consoles in the audio-for-video and film post-production markets, and we have assembled a team of highly skilled and talented personnel to continue to lead the industry in customer service and support."

Rick Plusher has been named as western regional manager, and will be leading the West Coast team, coordinating the Western Region sales staff, technical services and administration. He joins Neve after serving as president of Audio Intervisual Design, Los Angeles, and six years with Sony Corporation.

Laurel Cash has been named as a new western regional sales engineer; most recently she served as *R-e/p*'s advertising manager and, prior to that, was with JBL, Inc.

Michael George has also been named as western regional sales engi-

neer, a position he takes up after a career as an independent recording engineer and sales rep. Having worked for Westlake Audio, Los Angeles, in sales, he then went on to engineer projects for artists such as Leo Sayer, Talking Heads, Go-Go's and Lynda Carter, as well as many live recordings, such as the US Festival.

At the company's corporate headquarters, **Rhonda Kohler** has been appointed advertising promotions manager; she served formerly s *R-e/p*'s advertising sales special projects manager. Her duties at Neve will include advertising and promotional activites, as well as coordinating market research and customer relations.

Also in Bethel, **Gregory Davis** has joined the Technical Services staff, and will be involved in console commissioning and technical service. Prior to joining Neve, he served as VP of Manufacturing for Neotek, and has worked previously for Harrison, Inc.

NEW PRO-AUDIO REP FIRM FOR FLORIDA

Electronic Systems Marketing, Inc. is a newly-formed rep company that has been created to give sales and marketing support to manufacturers of pro-audio quipment throughout the state of Florida. The three principles of ESM are: Ron Neilson, Phil Dunkle and Martin Collins. Neilson has an extensive
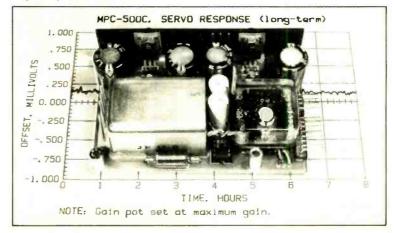
STOP PRESS: Cherokee Studios, Hollywood, has re-opened Studio One following the installation of a custom console based around vintage Trident A-Range mike preamps and equalizer sections. As can be seen from the accompanying photograph, the board features in-line buss monitoring, thereby doing away with the need for a meter bridge that can adversely affect the acoustics of a control room.

A feature article detailing the console's design parameters and topography, written by Denis Degher, will appear in the June issue of *R-e/p*.

EXPOSING AUDIO MYTHOLOGY

Laying to Rest Some of the Pro-Audio Industry's More Obvious "Old Wives' Tales"

by John H. Roberts

This month I will dust off my crystal ball and try to look at where we all are headed. In Part Two, we will consider some more performance specifications.

Towards The Future

Everyone in our business is trying to foresee (or *be*) the next trend in studio technology — that part of my crystal ball is already worn smooth. To make this a little more interesting, let's look 10 and 20 years off.

The dust on my crystal ball's window isn't too bad. When looking forward, however, it is often instructive (and much easier) to look back 10 years, and see how fast things are moving. In the mid Seventies, digital audio processing in the studio was rare, expensive, and barely studio quality (remember that some contemporary professionals still don't like the sound of Compact Disc). Those early digital devices were used for simple time delay, pitch shifting, crude

artificial reverb, and some experimental mastering/recording. The typical large studio of that era had between a third and half the number of tracks now available, while the smaller studio had a sixth to a quarter of its present complement. Today's typical home or personal-use studio simply didn't exist.

While there will be some acceleration in the rate of change, the studio business still moves relatively slowly. Because of competitive forces, the big (read: "expensive") studios spend more and benefit sooner from new technology than smaller studios. A recent trend that will pressure smaller studios to follow the big studios into newer technologies is the rapid growth of personal multitrack hardware.

WHAT'S IN A NAME?

Our apologies to *Rob Burr* for mistakenly referring to him as *Bob* on the title page of his feature article, "CD-ROM in the Studio," published in the February issue — *Editor.*

1996 ... And All That

With this factor in mind, let's take a look at what the mid-Nineties may look like. I expect the gap in the number of tracks between big and small studios to further narrow, as producers run out of ways to use all of those tracks. High-end digital recorders will move toward "soft" track formatting, allowing the engineer/producer to decide before each session how many full-range audio tracks, how many limited-range and/or data-compressed tracks, and how many automation/instrument control tracks he or she requires. Top-end machines will probably be able to mimic the track formats of older, fixed-track machines, so that tapes can be sweetened elsewhere.

Instrument and equipment automation will be pretty much standardized, probably using a variation on MIDI and/or IEEE-488 control busses. Many groups will come into the studio with their basic tracks already "in the can" — actually stored on a floppy disk, but which will look more like a business card than one of today's devices. The studio will be used to cut vocal and acoustic-instrument tracks; shooting performance video tracks; and sweetening. The full-service studio will not only prepare the (four-track?) digital master, but edit and assemble videos. Most big groups will release full-length video albums, while smaller groups can only afford to cull out one or two video singles, while hoping for a hit. Large stu-

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dios will be working with digital video, while the small guys will be stuck with crude old analog. Purists will begin complaining that digital video and the new wall size, liquid-crystal monitors are somehow "unnatural," and stick with their old-fashioned, CRT-based sets.

20 Years From Now

The liquid crystal balls we'll be using in 2006 will run much longer and cooler than my trusty old discrete crystal model. Unfortunately, my batteries are getting weak and those special clear see (through) cells are hard to come by here in Mississippi. So forgive me if the picture gets even fuzzier.

Not only will the marriage of audio and video be complete 20 years from now, but they will also be married to computer. Recording and playback will just be another form of data processing. Recording studios will almost certainly be equipped with very powerful, specialized computers (probably based on Reduced Instruction Set Computation, or RISC, and Transputer technology). Extensive communication capability will allow session players to 'phone in their gig. Near real-time jams between players that are not in the same studio also will be possible. A slight time delay will keep all the musicians from playing off each other in true real-time, but the drummer or another strong player could be designated the master with everyone

playing off him. The finished tracks can then re-assembled and synchronized together. While this might be a little rough for generating basic tracks, it'll sure beat flying back to L.A. to recut a guitar solo that mysteriously disappeared during mixdown.

The Studio of 2006 will look like a cross between a film-sound re-recording and a computer center; very little actual recording will happen there. Some larger studios will have sound stages and recording booths on the premises, but most musicians will cut their basic tracks at home. Extensive processing will allow acoustic spaces to be matched and enhanced. Basic tracks will be recorded very dry, since it is much harder to remove reverberation than add it.

The musician will be able to do much sweetening in his personal studio using his home computer — a small computer can do anything a big computer can do, just slower, by working off-line rather than in real time.

CAVEAT: Any resemblance between these predictions and what actually happens will be pure luck.

The Trouble With Squarewaves and Sinewaves

As we are all to often reminded (see the February 1986 and this issue's Letters page) care must be taken when trying to evaluate audio gear with artificially generated (non-musical) test

signals.

While equipment that measures well and sounds bad makes great fodder for the "Golden Ear" press, it is almost as easy to find equipment that measures badly and sounds good. The most obvious example of a test too rigorous for its own good is the ever popular squarewave. This presumably simple waveform contains harmonics going as high as the rise time of the signal generator.

As a circuit designer, I find it interesting to examine the way in which a given circuit deals with these out-of-band signals. However, such investigations have little relation to how that equipment will respond to music signals which don't extend that high. In theory, it is possible for a synthesizer to put out a true squarewave, but I doubt any will argue over the faithfulness of such a waveform's reproduction.

Wide-band squarewaves are difficult for any amplifier to handle, but negative-feedback amplifiers tend to screw up in visibly unpleasant ways. Various techniques are available to the circuit designer to attack the problem from different directions, although they share the common result of harmlessly attenuating the out-of-band components.

While I advise circuit designers to try and make their squarewaves look good, I caution equipment purchasers against reading too much into them. I know of

COMBINING ART AND TECHNOLOGY

The TS24 is the first in-line console from Soundcraft. And it represents a major breakthrough in in-line technology, because it now makes the console far easier to understand and operate.

Believe us, this is no hollow promise. Our argument is built around two rock solid foundations. Firstly, a new concept in console layout so logical, engineers used to split or in-line consoles can start work from day one. And secondly, a set of master conditions so advanced they'll amaze you.

STATUS.

One touch of the status button will configure the whole console for each particular stage of recording, mixing, broadcasting and video post production without sacrificing any flexibility whatsoever. In other words, one touch and you're off and running.

NEW DESIGN.

Conventional in-line consoles suffer from the limitations of one long travel fader and one equaliser being shared by two signal paths. With the engineer fader reversing and moving the equaliser back and forth throughout the recording, overdubbing and mixing process to optimise the situation.

The TS24 eliminates these short-comings, thanks to its logical design. The long travel fader is in the section called MIX, which is the signal path for both monitoring and mixing. The equaliser moves between the MIX and CHANNEL

signal paths automatically by use of the master status switches. 'Soft' switches may locally move EQ and AUX sends between the two signal paths but are also automatically reset.

When mixing, the Channel sections become available as additional inputs or effects sends without the limitations imposed by more conventional designs.

DROP-IN. BOUNCE.

Drop-ins are made easy by the use of the TAPE and GROUP button (T & G). Tape and Group enables you and the musician to monitor the original track and the overdub simultaneously.

The Bounce button facility enables you to take any combination of channels with their fader and pan settings directly to the routing matrix giving you instant bounce down.

SOUND AND VISION.

To create perfect sound, you also need perfect vision. With the TS24, that's exactly what you get. Separate scribble strips are provided instead of the usual confusing double one, and the Mix and Channel controls are in clearly defined areas for easier use.

AUTOMATION.

Soundcraft have developed a unique interface to the disc based MASTER MIX automation system, which enhances its operational flexibility by totally integrating the full extent of the console muting.

One feature of this system enables you to by-pass the Channel VCAs, thereby

optimising the original recording quality.
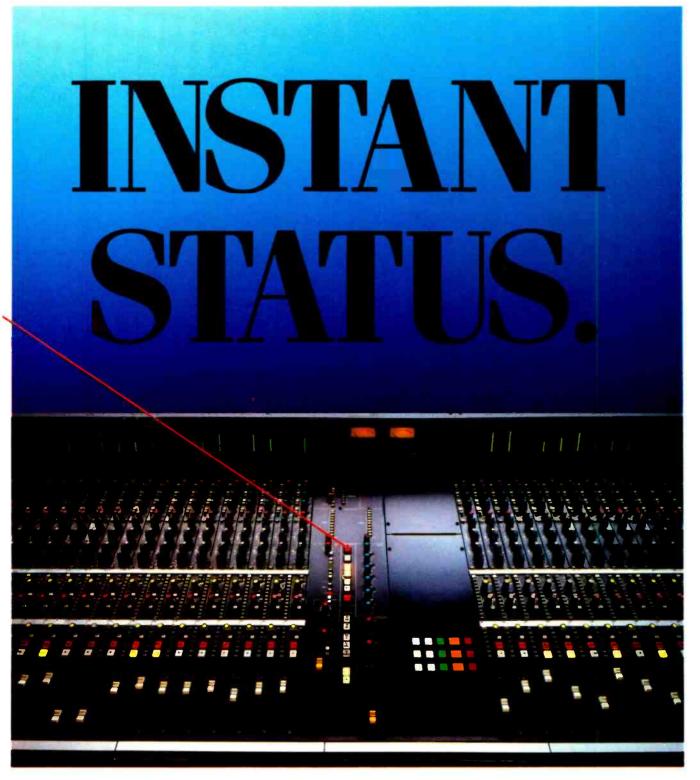
Surprisingly enough, all this practical technology, combined with sleek good looks doesn't carry a huge price tag. So our doors are open to practically everybody.

Which only leaves us with one thing to say: if you want to keep your finger on the button in the most up-to-date mixing console design available, contact us.

Soundcraft TS24

INSTANT STATUS.

MARSHALL MORGAN
Producing a String of
Country Hits

Interviewed by Bruce Borgerson

Color photography by Beth Gwinn

As engineer/co-producer for Ricky Skaggs, The Nitty Gritty Dirt Band, The Whites, and Lacy J. Dalton, among others, Marshall Morgan has emerged as a pivotal figure in country music's "Neo-Traditionalist Revolution" — a movement often blending doses of rock 'n' roll energy with pure country and bluegrass.

Morgan, a mid-Tennessee native, began his music-industry career during high school and college, working on-stage as a guitarist and keyboard player, and off-stage as a sound mixer. (One college band included future production partner Paul Worley.) After receiving a degree in English from Vanderbilt University, Morgan went to work for Carlo Sound, a Nashville concert-sound contractor. Quickly assigned to an Eagles tour, he struck up a friendship with the California rockers, and soon joined their road crew. After three years with the Eagles organization, Morgan returned to Nashville to build a career as an engineer and producer.

With the exception of Ricky Skaggs projects, Morgan currently only engineers sessions on which he also plays a production role. In addition, he is scheduling his outside demands to allow more time for songwriting, a creative endeavor often done in collaboration with successful singer/songwriter (and wife) Nancy Montgomery.

R-e/p caught up with the busy engineer/producer at Audio Media Recorders, Nashville, between overdub sessions for a forthcoming Nitty Gritty Dirt Band album.

R-e/p (Bruce Borgerson): This is your third album for the Dirt Band. Are you taking any kind of different approach with the band this time?
Marshall Morgan: Yes, the way we're cutting the Dirt Band this time they're playing all their own instruments, which is a lot of fun. It gets away from the normal 10, 2 and 6 Nashville session grind — which sometimes can be great because there are *so many* superb musicians here. At its best, the Nashville "Session System" is wonderful but, at its worst, it can turn into a formulaic way of recording! So this album was an opportunity to do a self-contained

type of thing, where we rehearsed for two weeks before cutting tracks. We used Treasure Isle [Nashville] to get more of a live sound; more room ambience than is available here at Audio Media.

R-e/p (Bruce Borgerson): Do you have a title for the album yet?
Marshall Morgan: Not really, though [manager] Chuck [Morris] mentioned a working title, *No Excuses.* It's going to be a good one; we're excited about it. We're getting some creative, spontaneous playing on it. For me, spontaneity doesn't come from *not* knowing what you're

going to do, and playing off the top of your head. But rather from rehearsing to the point where you know *everything* you're going to do, and then just *doing* it, instead of thinking about it or reading it off a chart; it's a sort of "Zen Concept." By doing that, we're getting some fresh, creative, good-feeling musical things that you might not get in a studio musician atmosphere.

R-e/p (Bruce Borgerson): Do you plan to mix the new album here as usual?
Marshall Morgan: I'm not sure yet. A new console is in the works for this room, and I hope it will arrive before I

MARSHALL MORGAN
— Country Engineer/Producer —

get to the mixing. Otherwise, I will mix elsewhere. I think we'll go with the Trident 80B for this room, which is a relatively inexpensive console, and invest the extra money in a lot of outboard gear.

R-e/p: How you make such great records using these "non-standard" monitors? You don't have the obligatory Kleenex over the tweeters on your Yamaha NS10s.
MM: I don't feel I need the Kleenex on these because they are an older pair, and the top-end isn't quite as bright as the newer ones. Yamaha swears there's no difference, but if you go out and buy a new pair of 10s, and set them side by side with older ones, they sound quite *different*. We investigated and found out the woofer and the tweeter are the same, but the cross-over is different in the newer ones. I think what happened is that Yamaha had trouble with people blowing tweeters on the originals, so they crossed it over to keep out more bottom. At least that's what we measured, though Yamaha swears we're crazy. [Readers might like to refer to an article by Bob Hodas, entitled "Examining the Yamaha NS-10M 'Tissue Paper' Phenomenon," published in the February 1986 issue — *Editor.*]

R-e/p: How long ago did they change the crossover frequency?
MM: Oh, at least a couple years. I've been using NS10s since about 1980, when Kyle Lehning turned me on to them. I've also been experimenting with the little B&W Model 100s, though the jury is still out on those as far as I am concerned. They are not as bright as the Yamahas, so they make you work a little harder, which is good.

R-e/p: Do you rely on your small monitors, or the big room monitors?
MM: I split about 50-50 for tracking and mixing. On overdubs, I'll stick mostly to the little ones to minimize ear fatigue.

R-e/p: How much of a part did you have in the selection of the big monitors here at Audio Media?

Plain Dirt Fashion

MM: Quite a lot. These are — well, they used to be — Westlakes; they have the basic Westlake horn with JBL woofers and TAD drivers set up in a two-way system.

R-e/p: What is that little black box you have on the console with two faders in it?
MM: That's my fader package for doing overdubs. I don't run anything through this old console anymore; we cut some good things on it, but it has definitely seen its better days. At this point, I use it as a monitoring device only. On this new Nitty Gritty Dirt Band album, I'm cutting tracks out at Treasure Isle, using their Trident Series 80 and outboard Trident A-Range modules. For overdubs here, I'm using a Sontec MB-1 pre-amp and this fader package made for me by Morris West, a guy here in Nashville who makes a lot of guitar gear. It has one 5532 op-amp and a P&G fader for each channel, and that's it. It's very simple, very clean, with 5 dB of gain if you need it. I also have an API EQ rack here, which I use along with the LA-2A for vocals. It makes for a very straight signal path with a lot of transients. Of course, I can only do two channels at a time, but for overdubs that's usually all I need. I've experimented with different pre-amps for a couple years, and this is the first I've found that I really like for a wide variety of things.

R-e/p: Is this the first time you've cut tracks out at Treasure Isle?
MM: Yes, and I like it there. I like the Trident, which is very good sounding basic console if you have the right outboard EQ and signal processing. They also have 12 of those A-Range modules, with pre-amps and EQ sec-

tions, accessible immediately to your right so you can have enough channels for practically everything. It's great when you need a lot of punch — for kick or snare, for example. I normally use all 12 modules.

R-e/p: How would you compare the Trident A-Range EQ to your API rack?
MM: The API is a "harder" EQ; it's very "crisp" sounding. The Trident is very "silky," with a lot of gain on the top-end. You can really crank it up a lot without it becoming offensive; it's a very warm, flexible EQ. Technically, I don't know why. It just *is!*

R-e/p: There's a 3M DMS digital multitrack in residence at Treasure Isle, yet you're doing this album analog. Why's that?
MM: Budget restrictions were part of the reason, though we probably could have squeezed it in if we really wanted to. But I didn't push for it because, they way we are doing this album, we do a lot of editing between takes to get the best nuances and the best feel. With the 3M, you can't do razor-blade edits, so you need two machines. Whereas with analog, I can splice things together on the spot, and see if I've got what I need. Also, because we're doing most of the overdubs here, we would have to haul the machine back and forth, or rent another one. I've found that the 3Ms don't transport well; they prefer to stay put.

R-e/p: I assume that you're referring to experiences with the new Ricky Skaggs Live in London *album that came out late last year. Why did you go all the way to London to record those sessions?*
MM: I'm not really sure. Ricky just wanted to expand his horizons and, I guess, to do something different.

R-e/p: What was behind the decision to record on digital multitrack which is, I believe, a first for a live country album?
MM: Digital to me is a terrific medium. In general, digital multitrack is a joy to work with, and a real time saver. For live recording, it's a natural, because during a show you will have wide variation in levels, from a hot up-tempo to a slow, quiet ballad. If you are not using noise reduction — which I personally can't

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"For me, spontaneity doesn't come from *not* knowing what you're going to do . . . but rather from rehearsing to the point where you know *everything* you're going to do, and then just do it, instead of . . . reading it off a chart."

MARSHALL MORGAN
— Country Engineer/Producer —
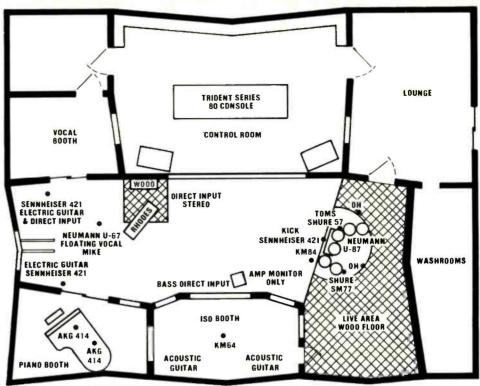
stand — then you're working your tail off just to keep your levels even, unless you compress everything, which I also don't want to do. The only other route is to rehearse the recording a lot more than we would have time to.

Digital is very forgiving in terms of level. If it's a little bit cold on the tape, that's okay, because you don't pick up any more tape hiss. It also allows you to use all the natural dynamics of a live-recording situation, without a lot of level riding. We are *very* pleased with it, and I think when it comes out on Compact Disc, you'll really notice it. It has a *lot* of punch!

The selection of 3M machines was a joint decision by Ricky and myself. We had both listened to the 3M and some others by sneaking around studios here in Nashville, when others were using them. We liked the 3M the best, especially on the bottom-end which is solid *and* punchy.

R-e/p: How did you go about arranging rental of the recording truck you used in London and the digital multitracks?

MM: We chose the RAK Mobile because it has an old API console: a simple, standard warhorse console that passes a good signal. I also like API EQ — I have a rack of their equalizers here, and I knew that would allow me to easily duplicate the EQ on any necessary "fixes." It was pretty crowded inside the van, since it was just a little 24-foot truck — or "lorry" as they say over there. We had a pair of the 3M digital multitracks that

Frank Dickinson[president of Digital by Dickinson] took over; he was instrumental in putting this whole project together. We had them set up in a separate van, which was backed up to the RAK truck since there was no way we could fit them in — it was just a bare truck with lights taped to the ceiling. We had to jack it up on hydraulic jacks to get it exactly level, so the machines wouldn't roll away! It took Frank hours to get the machines set up and aligned for each gig. They're very stable once they get going, but they tend to go out of

whack if you turn them off and transport them around.

We mixed back to two tracks on the 3M, which was one of the greatest revelations of my recording career. It was wonderful! If you don't like something on your mix, just go back and punch-in. No sweat, and *totally* undetectable! I was punching-in words on the actual mix. I did things in the final mix that I would normally do earlier on a vocal comp. It took all the pressure out of mixing, and made it a complete joy.

We recorded three shows, but almost the entire album was taken from the

ROOM LAYOUT FOR NITTY GRITTY DIRT BAND SESSIONS AT TREASURE ISLE, NASHVILLE

Recent Nitty Gritty Dirt Band session (*standing*): band members Bob Carpenter, Jeff Hanna, Jimmie Fadden, John McEuen, Jim Ed Norman (Executive VP, Warner Bros. Nashville), and Jimmy Ibbotson; plus Chuck Morris, Marshall Morgan and Paul Worley.

Melodie Gimple

Partial Discography

Engineer and Co-Producer (with Paul Worley):
Nitty Gritty Dirt Band: *Partners, Brothers and Friends,* and *Plain Dirt Fashion.*
Lacy J. Dalton: *Can't Run Away From Your Heart.*
Gary Morris: *Gary Morris*

Engineer and Co-Producer (with Ricky Skaggs):
The Whites: *Whole New World*

Engineer:
Ricky Skaggs: *Waiting for the Sun to Shine, Highways and Heartaches, Don't Cheat in Our Hometown, Country Boy,* and *Live in London.*
The Whites: *Old Familiar Feeling,* and *Forever You.*
Anne Murray: *A Little Good News*
Mickey Gilley: *That's All That Matters, You Don't Know Me, Fool For Your Love,* and *Put Your Dreams Away.*
Johnny Lee: *Sounds Like Love.*
England Dan and John Ford Coley: *Dowdy Ferry Road,* and *Some Things Don't Come Easy.*

> **"If Paul Worley and I have found
> a trademark sound, I guess it would be
> that lean, 'less-is-more' approach,
> particularly when compared to the excesses
> of the so-called 'Nashville Sound'."**

MARSHALL MORGAN
— Country Engineer/Producer —

last night at the Dominion Theater in London.

R-e/p: Whose microphone selection was used for the recording?
MM: I pretty much picked them. We used a combination of those from the sound company and the RAK Mobile; the vocal mikes were the same as those used for the PA. I picked them in consultation with Ricky's live engineer, Phil Somens, who does a superb job for Ricky. Nine times out of 10, our choices were the same. I think we used some different mikes on acoustic instruments, and on the piano.

R-e/p: What's that old vocal mike Ricky uses on stage?
MM: Aha, that's an interesting device! The shell is an RCA 77, but the guts of it are an Electro-Voice PL95. The shell is really just a prop, though we have used the original 77 in the studio for other things. They don't have a whole lot of top-end; it's a ribbon mike that rolls off about 7 kHz. It works great on banjo, because it makes it sound like Earl Scruggs. Putting an E-V capsule in was Phil's idea. Ricky liked the look of it, but it didn't have enough top-end for vocal.

We used up about 26 tracks on the 3M during the recording, give or take a few, depending on who's playing what, and that's not counting four tracks of various timecode.

R-e/p: The recording makes the Dominion Theater sound rather small and dead. Do the acoustics actually sound like that?
MM: Well, to tell the truth, we had to keep the audience mikes way down in the mix, because we had to go back and fix a few vocals. Digital is *so* accurate that you could pick up the subtle difference between the live audience mike, and the "fixed" vocal in the studio. And, because we wanted a uniform sound throughout the album, we had to keep the room sound down throughout, and replace it with studio echo — which, of course, I don't like to use much of in any case.

R-e/p: Did you mix it here at Audio Media?
MM: Yes. We brought two 3Ms down

to do editing, and had to put them out in the hall at that stage. When we got to the stage of mixing with just one machine, we brought it into the control room.

R-e/p: Were the same mixes used for the BBC TV Special, or was that mix done separately?
MM: No, those are the same mixes. As it turned out, they liked the same performances we did. We did some extra mixes just for the video, which will be released as a Home Video here as well.

R-e/p: Did you mix the video?
MM: No, though I did check it afterwards to make sure any "fixing" we did was still in sync. They essentially cut the video into the audio.

The video was done independently by Sandy Fullerton out of Los Angeles, so all the post work was done out there. But the first sale was to BBC: it was assembled in its first form for them, and first shown over there.

R-e/p: Will the next Ricky Skaggs studio album be all-digital as well?
MM: Yes, definitely.

R-e/p: Are you talking to the owners of Audio Media about buying a digital multitrack?
MM: No, we'll probably rent again, though I'm thinking about buying one for myself — if I only knew what to buy! I'm reluctant to sink 150-grand into something, and have it become obsolete next year.

R-e/p: You were one of the first engineers in Nashville to start mixing to digital two-track. Do you remember the first album you remixed digitally?
MM: One of the Mickey Gilley records, I think. I'd have to look on the credits to tell for sure. I know *Highways and Heartaches* was the first one for Ricky Skaggs, so I guess I've been doing it for around four years.

R-e/p: So Ricky Skaggs' Waiting for the Sun to Shine was an analog mix? It's still a great sounding record!
MM: As a matter of fact, that project was mixed to quarter-inch on that old Studer B67 sitting over there, which we now use only for rough mixes and echo. What's more, the tracks were done on an old A80 multitrack at 15 ips with Dolby. Ricky wanted to use

that format to save on tape costs, because the CBS deal hadn't gone through yet. What saved us is the fact that we never left this room, so we didn't have to worry about Dolby alignment.

R-e/p: Back to digital. It seems that Nashville began switching over to digital mixing much more quickly than either New York or L.A. — opposite to the usual trend. What was the reason for that?
MM: I think Glenn Meadows [of Masterfonics and Master Technologies] had a lot to do with it. He took a chance and bought several of the JVC [BP-90 PCM processors], and made them readily available at a reasonable cost. I think many people here felt the improved results warranted the extra expense. Because it's a small recording community here in Nashville, word spreads fast.

R-e/p: Let's jump back in time, and talk about how you started working with Ricky Skaggs.
MM: I first met Ricky when he sang harmony on a Janie Fricke record I was engineering for Jim Ed Norman. I had never really heard anything about him before that. This was while he was still on Sugar Hill; before the CBS deal. I asked Jim Ed about him, and found out that Ricky was a highly respected multi-instrumentalist who had just left Emmylou Harris' band and moved to Nashville.

Well, at the same time we were doing this Cynthia Clawson gospel album, one number needed a bluegrass-kind of sound. We had done the basic tracks, so I asked Ricky to come in to overdub guitar, mandolin, fiddle, and banjo — he did a whole number on it, even singing harmony! We enjoyed working together and I guess he liked my engineering. At the time he was working on his CBS deal, so he called me to do *Waiting for the Sun to Shine*. That was all cut before the contracts were signed, but that's Ricky — confident all the way!

MARSHALL MORGAN
— Country Engineer/Producer —

R-e p: Did you consciously have a new sound in mind for that album? It was obviously a different approach — not really country, not really pure bluegrass, and with a strong rock influence in the rhythm tracks.
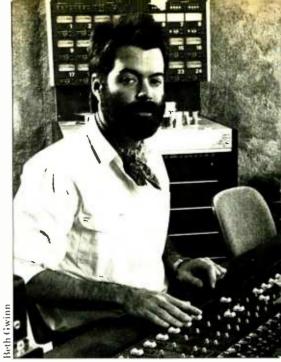
MM: From an engineering standpoint, it wasn't so much what I *did* as what I *didn't* do: I just kept it pure and simple. What makes the "Ricky Skaggs Sound" is more in the choice of instruments and the material. But I was definitely aware that we had a new, fresh approach to things, although I didn't really change my techniques significantly. I've always seen myself as an engineer who, in a sense, puts music first and technology second. I'd had a lot of experience with acoustic instruments, and knew how to bring out a strong, natural sound — an "uncluttered" sound. I don't use a lot of echo; I guess you could say I'm a minimalist engineer. Anyway, it just fell together with what Ricky was doing.

R-e p: What about that prominent bass/drum sound on the Skaggs records? Who is responsible for that?

MM: It was Ricky's idea to have the bass and drums on bluegrass material, but the overall sound was mine. I'm *drum* oriented — I've been accused of having the drums louder than the vocals — sheer *heresy* in country! In my early days I was a snare-drum fanatic, though I hope I had that in proper perspective by the time I started with Ricky.

R-e p: Did you have any hunch that Waiting for the Sun to Shine *would take off the way it did?*

MM: I didn't have any inkling it would take off so fast. We all felt good about it at the time, but not many country artists sell 800,000 on their first album! Ricky is a confident guy,

and thought he would be successful, but I don't think any of us knew what would follow.

R-e'p: On one album jacket, Skaggs referred to himself as "picky Ricky." Is he a demanding artist and producer to work with?

MM: He's very demanding of himself, and of the quality of the music. But, when he met me, I think he met his match when it comes to pickiness. He and I pretty much strive for the same goals. Also, since we've both been successful producers for the same length of time, we have respect for each other.

R-e/p: Since Ricky is his own producer, do you feel any conflict, any urge to slip over into the producer's chair — which is, of course, empty a lot of the time?

MM: I do play a limited producer's role with Ricky, but only in the sense of execution. I don't have anything to do with pre-production, with finding songs and working on arrangements; Ricky does that. But, when it comes down to actually executing what he has conceived, I play a co-production role. He has to do so much himself: he plays so many instrumental parts, and does most of the harmony singing himself. He needs somebody in the booth to say "yea" or "nay."

R-e/p: How involved does Ricky become in mixing stage?

MM: Quite a bit. I'll work up a mix for a couple hours, then he'll come in and take over while I take a break to rest my ears. What he does is primarily work with the harmony vocals, fine-tuning them until he hears what he likes. He'll fine tune the EQ and compression along with the levels.

...continued overleaf —

RECORDING ACOUSTIC INSTRUMENTS:
Marshall Morgan's Techniques for Ricky Scaggs Session

Since recordings that have been engineered by Marshall Morgan are noted for their crisp, clean sound on acoustic instruments, *R-e/p* asked him to explain his recording techniques.

- **Acoustic Guitar:** "What I do with Ricky, since he has a number of acoustic guitars, is to start by trying different ones. For mikes, I often use the Neumann KM64, the older tube version of the KM84. Another interesting one is the Sony C37A. We'll change around guitars and mikes to differentiate sounds, especially if we are doing a stereo pair. I also fool around with acoustic environments, using a rug or taking it away, using the hard side or soft side of a baffle.

  "I'll sometimes use a little compression with a [UREI/Universal Audio] LA2A to hold level. I use quite a bit of high EQ; how much depends on how new the strings are. On the first few passes, I may need only +4 at 10k, but after about four takes — if he hasn't gotten his part — you start cranking it up. On my API, for example, I'll often end up at +9 on the 10k shelf, with maybe +2 at 5k. I use outboard equalizers almost exclusively. Usually, I'll rent additional EQ for tracking or mixing — a Sphere EQ sounds nice on some things, for example. For overdubs, I stick to the studio Pultec and my APIs."

- **Banjo:** "That varies, depending on a great deal of what it is competing with in the sound spectrum. If you have bright acoustic guitars, I'll go for a medium range to cut through. For that, an RCA 44 or 77 ribbon does very well, though I've also used KM64s and [Neumann/Telefunken] M49s on banjo. I probably change mikes on a banjo more than any other instrument, sometimes trying three or four before I find one I like. Sometimes I'll do strange EQ, like at 1.5 or 2k. Occasionally, I'll add top-end, like on guitar, but you have to be careful because than can become 'offensive' with banjo."

- **Mandolin:** "Usually a KM64; it seems to be a good all-around acoustic mike. That was something I picked up from Bill Szymczyk, though he usually used the newer KM84 transistor version. Mandolin EQ is similar to acoustic guitar.

- **Dobro:** "I'll often use a [Neumann] 249 on dobro; it has a relatively large pattern which can pick up the whole instrument. Using other mikes with narrower patterns can make placement too critical. With dobro, the more 'overview' I get the better it sounds, so an M49 or 249 works well for that. I'll use light compression and high EQ, though not as much as guitar, maybe only +4 at 10k. I try not to EQ in the midrange at all. That's where you can make things sound unnatural."

- **Fiddle:** "I'll usually use a 249 or KM64, depending on the instrument and the song. I'll mike it a foot away, minimum, more like 18 inches usually. With a KM64 you have to *make* sure the fiddle player stays put — with the pickup pattern aimed right at the body of the fiddle — and not back toward the tailpiece, because that'll give shrill overtones."

- **Piano:** "Most often a pair of AKG 414s, sometimes Schoeps SK5s. I think the sound of the piano is the most important thing."

- **Instrument Reverberation:** "I prefer an old EMT plate, which usually gives the warmest sound. I find that some digital reverb often doesn't react well to high frequencies. I usually use very little reverb; acoustics are pretty *dry* in my mixes." □□□

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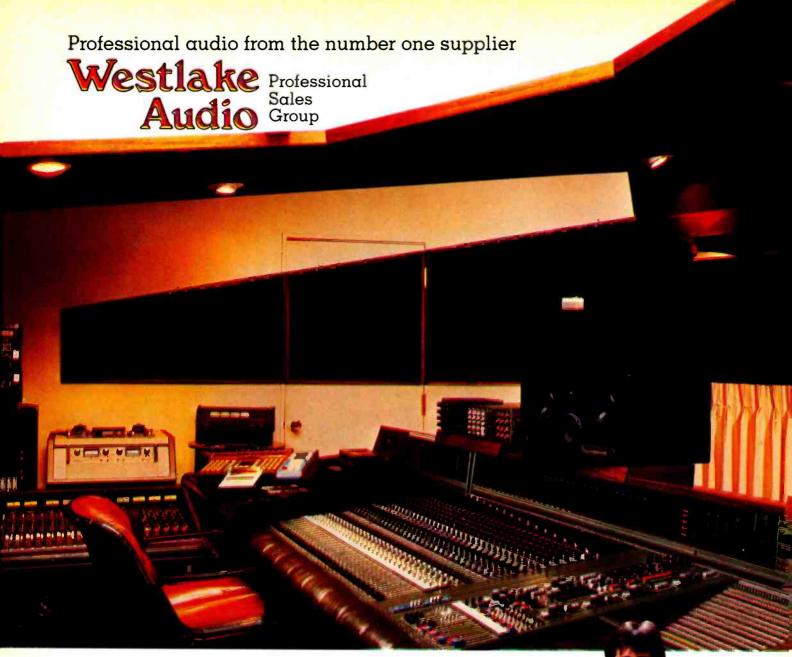
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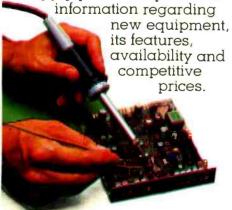
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MARSHALL MORGAN
— Country Engineer/Producer —

R-e/p: Does he work with a second engineer to assist on that?

MM: No, he does it by himself — I've trained him pretty well! But seriously, he did have some engineering experience previously, working in a small studio in Kentucky. There are even times when he'll engineer overdubs by other musicians. But he doesn't change anything other than harmony vocals without asking me. And, since we don't use automation, he will help me on the final mix.

R-e/p: Have you done many automated mixes with other artists?

MM: A few, but I don't feel I need it. For the kind of records I do, it takes more time to set it up than it does to just go ahead and mix the song. It also has to do with the way I record, since I'm extremely level conscious. I do a lot of riding on tracks, and when I'm pinging to put together a comp vocal. By the time the tracks are ready, the songs are pretty close to mixing themselves!

R-e/p: Let's plunge back into the mists of time and talk about your involvement with the Eagles. When did that begin?

MM: That was 1973 through '75. I started with them just when they'd come back from England, where they'd done *Desperado* with Glyn Johns. It wasn't a very successful album commercially at the time, although it was a great record — a classic. I worked through the *One of These Nights* album, which was the breakthrough album that launched them up to multi-platinum status. I worked through that tour, and quit to move back to Nashville just before *Hotel California.*

R-e/p: What was your function with the Eagles?

MM: At one time or another, I did just about everything for them: I started at monitor mixer; juggled the guitars on stage; and, in a pinch, did house mixing. This was the early days when there were only three people in the entire road organization. They had a road manager who herded the band around, then there were two of us who did everything else: handled the equipment, drove the truck in, set-up the stage, did the gigs, took it down, and drove to the next place. Later, I wound up being their production manager, and was pretty much responsible for the production of the entire live show.

When I started we were doing small venues — gyms, cafeterias and theaters at small colleges, up to two and three-thousand seats. Of course, by

the last tour we were up to the 15- and 20-thousand seat coliseums and stadiums. We opened some shows for the Rolling Stones one summer, so I had the opportunity to see it go from small change to megabucks. It was a lot of fun.
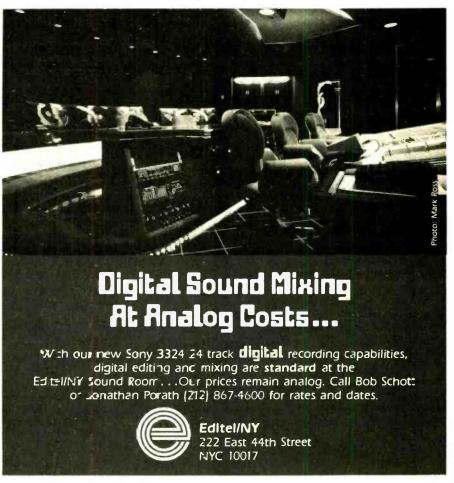
R-e/p: Did you spend much time in the studio with the Eagles?

MM: Yes, I was with them through most of *On the Border*, and through all of *One of These Nights.* I was basically playing nursemaid, getting the band to the studio, and helping them with the instruments. Since I was pretty much always there, one of my formative studio influences was watching [session producer] Bill Szymczyk at work. He was another guy who, as far as I know, did not have a lot of formal musical training. But he did have a great seat-of-the-pants feel for what was good — for when a track was hot, and when it was not. The band always called him "coach." He was a great motivator; a guy who could say the right thing at the right time.

He also had a lot of patience. Recording with any band like the Eagles demands patience. There was a lot of cutting things over and over, splicing tapes together — things you don't need to do as much when working with studio musicians. But, on the other hand, with a band you usually get more excitement, more unpredictability, and that was part of what made the Eagles great. Bill made them sound slick enough, but not *too* slick. Yeah, Bill was a strong influence on me, a great talent!

R-e/p: Although I've never met Szymczyk, I once talked to a second

MARSHALL MORGAN
— Country Engineer/Producer —

engineer he worked with at Criteria, who said that Bill does an incredible amount of takes and splicing together of tapes.

MM: True. The band played the song "One of These Nights" for two days and two nights, almost straight through. I'm not sure just what got on the final version, but I know it involved multiple tapes. They must have finally got it from sheer exhaustion!

R-e/p: Did you sit in on any of the mixes for the Eagles albums?

MM: Not in any participatory function, just as an onlooker, going in and out. I wouldn't say I was able to pick up any specific engineering tips from Bill, since I had no studio experience at the time. What I did get from him was an *attitude*: a vision of the producer/engineer's potential role can be. He showed me what you could accomplish sitting in that chair.

R-e/p: When did you move into the studio recording side of the business?

MM: At the end of 1975, I pretty much decided that I'd had enough of the road. I had a good job, but all I could see it leading to was more of the same. So I moved back to Nashville, and got

a job at Quadrafonic Sound Studio working with Gene Eichelberger, who was one of the more progressive engineers in town. He was one of the few who had worked with out-of-town pop artist: Neil Young on *Harvest*, and Dobie Grey on "Drift Away," among others. He had pretty much pioneered a pop attitude here in town.

Gene was also a German taskmaster who taught me some good, basic recording procedures. I worked there as a second for about six months, working pretty much exclusively with him. After that, I slipped in to the engineer's chair pretty quickly, since I wasn't exactly green; I had a lot of experience working with people, and a good ear for sound

Next I went out to Studio by the Pond, and worked with Kyle Lehning, who had just had his first hits with England Dan and John Ford Coley. He's another producer/engineer but, at that time, he wanted to get out of the engineering chair to concentrate on the musical aspects. It was a perfect opportunity for me, because I was doing the engineering while learning from him at the same time. He wanted somebody to do it pretty much his way, following his own style. I would cut all the tracks, do the overdubs, and then we would mix it together. We'd bounce back and forth, trading

ideas around.

R-e/p: When did you make the move over here to Audio Media?

MM: I was dabbling in producing while working at the Pond, and I ran into Paul [Worley] again one Christmas, and hired Paul and the Audio Media rhythm section to play on my project. Jack Jackson, the bass player who was also one of the studio owners, liked my engineering, and hired me as staff. Which was great, because the production died and my work with Kyle had run its course, primarily because he was doing more work out of L.A. So I took the job and, at the time, they were doing some fairly weird specialty product types of things: Christmas albums, jingles; sound-alike golden oldies; all different kinds of off-the-wall things like a series of albums for the National Geographic Society.

One thing I liked about these projects was cutting tracks live with a roomful of musicians — horns and strings and rhythm section live all at the same time. It made me appreciate the way they did it in the old days, cutting live to mono.

I was on staff there for less than a year, doing some independent production on the side, working with Paul as co-producer. Nothing happened right off, but we did get our teamwork going at that point. After a year I started cultivating freelance jobs, and eventually quit the staff job.

Well, to tell the truth, I got fired! For once I lost my patience with a client, a well-known songwriter, and that resulted in a breakdown with one of the four owners of the studio. But Audio Media was like home to me, so I came back as a client, and basically took it over for a good part of the next four years.

R-e/p: So they kicked you out the door and you've been there ever since?

MM: Right! You see, there had been no major-label hit albums out of the studio until I started working there. Since they saw this as a good source of future income, it put me in a position to demand some changes in the studio, supervising some upgrading to fine tune it to my liking. I basically worked with Jack Jackson to re-build the studio. We added outboard gear, improved the microphone selection, and changed the room around some, making it into a competitive music recording studio in my first years there.

R-e/p: What were the first hit records you cut there?

MM: The first was by Jeanne Pruett in 1978 or '79, but we didn't really get rolling until about a year later when

MARSHALL MORGAN
— Country Engineer/Producer —

Jim Ed Norman came to town. I had met Jim Ed when I was with the Eagles, because he had been doing all of their string charts. So I thought, "Hmm, this perhaps presents an opportunity." He had just done some tracks for the *Urban Cowboy* soundtrack, and was coming here to work with Mickey Gilley.

So I called him, went over and had a talk, and convinced him that he should hire *me* as an engineer, and use the Audio Media studio and rhythm section. I told him this was a good studio with a bunch of young guys who would be the top rhythm section in town in a year or so — especially if you gave us all this gig! Well, he did, though we did plug in Joe Osborn on bass about that time. We went in, did the sessions on Gilley, and everything we did went to Number One on the country charts — I think we had about a dozen Number Ones in a row. We also cut Johnny Lee and Janie Fricke.

Later, we would up doing Anne Murray's "A Little Good News" with Jim Ed — most of that was done at Audio Media. We also did Michael Martin Murphy there. Some of these were exclusively Nashville projects, and some were done partly in L.A., where Eric Prestidge usually handled the engineering.

Along about this time, I brought to Jim Ed's attention that Paul and I would like to move more into production, hoping he could steer something our way. So we became part of JEN Productions, and the first session he found for us was a gospel project, Cynthia Clawson. We had never done any gospel, but it went on to win a Dove Award as Inspirational Album of the Year. Also, about this time, Gary Morris approached me. I think he wanted to co-produce himself, and was looking for a producer/engineer to work with. I convinced him that it would be better for Paul and I to do it, and the first single we did with him went to Number Six — the first country single we every produced! We went on to have a good string of hits with Gary.

R-e/p: How did you and Paul Worley hook up with the Dirt Band?
MM: They had just finished an album with both Norbert Putnam and Richard Landis producing, but they were not really fully comfortable with either. Bruce Hinton, who was a partner of Jim Ed's at the time, suggested I do some editing and remixing on "Dance Little Jean" for single release, starting with the tracks done by Norbert for the album. So I did, and that

single turned out be very successful; they then asked Paul and me to try an album. We did *Plain Dirt Fashion*, which did very well on the air and in the stores.

R-e/p: Had you followed their careers over the years?
MM: Yes, as a matter of fact they did a show at Vanderbilt back in the early days, in the first phase of their career, and they were just incredible. It was of the *best* live shows I have ever seen; it was talked about for years. And then 13 years later, I wind up being their producer!

R-e/p: How did the two of you get the nod to produce Lacy J. Dalton?
MM: I believe the people at CBS suggested us. Lacy was going to work with Steve Buckingham, but he had to pass on it for other commitments; so Paul and I were actually second choice. I think Lacy is a tremendous talent. Her songwriting is good; I'd like to see her write a whole lot more. We did have a little trouble finding a direction for that album. We cut one thing that was very rock 'n' roll, but it was too far out in left field for the other things we had, so we had to set it aside. Lacy finally came through late in the game with some songs that helped put it together.

R-e/p: Was it intimidating to work with Lacy, considering it meant following in the footsteps of Billy Sherrill, the legendary country producer who discovered her, and produced all her previous albums?
MM: No, never thought about it. I have a lot of respect for Billy Sherrill, and he made some really good records

with her. But later on, she wanted to do some different things, and it wasn't what he was really good at. It was obviously time for a change.

R-e/p: Was the sparse, spare sound you captured on that album, Can't Run Away from Your Heart, a conscious move away from what she had been doing before?
MM: No, it was not a deliberate change. Paul and I don't really think of ourselves as having a "production style" as such; we try to adapt what we do to fit the needs of the artist. But if we do have a trademark sound, I guess it would be that lean, "less-is-more" approach, particularly when compared to the excesses of the so-called "Nashville Sound."

R-e/p: Whatever it is you do, to the best of my knowledge the critics seem to have unanimous praise for just about everything you've done.
MM: Yes, the critics have been great just about everywhere, but, unfortunately, they get their records for free! With Lacy, for example, the singles frankly haven't been doing as well as we had hoped. And, unfortunately, that's what you need to sell in country. I think her album holds up well as a cohesive work, though.

R-e/p: Besides writing hit songs with your wife, what does the future hold for Marshall Morgan?
MM: For one thing, I'm really interested in moving more into rock 'n' roll production. I'm really a rocker at heart, and have been involved in it for years. Unfortunately, rock acts are not looking for producers with a track record of country hits! ∎∎∎

Color photography courtesy of JBL Professional

NEIL DIAMOND ARENA TOUR

Stanal Sound On The Road With The New Concert Series Speaker System

by David Scheirman

First introduced to the public in 1985 by JBL Professional, the new Concert Series represents what is perhaps the first attempt by a major manufacturer to address the growing need for pre-manufactured speaker systems, complete with all accessories.

In recent years, several firms have introduced factory-built enclosures that have found acceptance in both the touring-sound and installation-contracting industries, and a few of them included accesories such as cables, dollies, and flying hardware. The significant development is that the Concert Series systems are complete and ready to operate with the addition of only program source and mixing equipment. As part of the new series, loudspeaker cabinets, power amplifiers, electronic crossover and loudspeaker signal-processing equipment are available, as well as equipment racks. The systems are pre-wired, tested and ready for immediate use, with road cases available for touring applications.
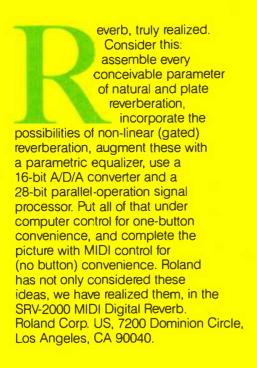
Typically, touring-sound firms have steadfastly clung to the development of proprietary loudspeaker systems, maintaining that the established manufacturers do not build what they require. Firms such as Meyer Sound Laboratories, Eastern Acoustic Works and Turbosound have been changing that. Each of these companies has made significant inroads into the touring-sound market with mass-produced enclosures. The new Concert Series systems, which come complete with power amplifiers and signal processing, add a new dimension to the Pro-Sound marketplace, with potential applications in installation contracting, as well as the rental-sound business.

This article will examine the development of these new JBL systems, and their first large-scale use by Stanal Sound for Neil Diamond's recent arena concert tour.

Touring Sound Company Involvement
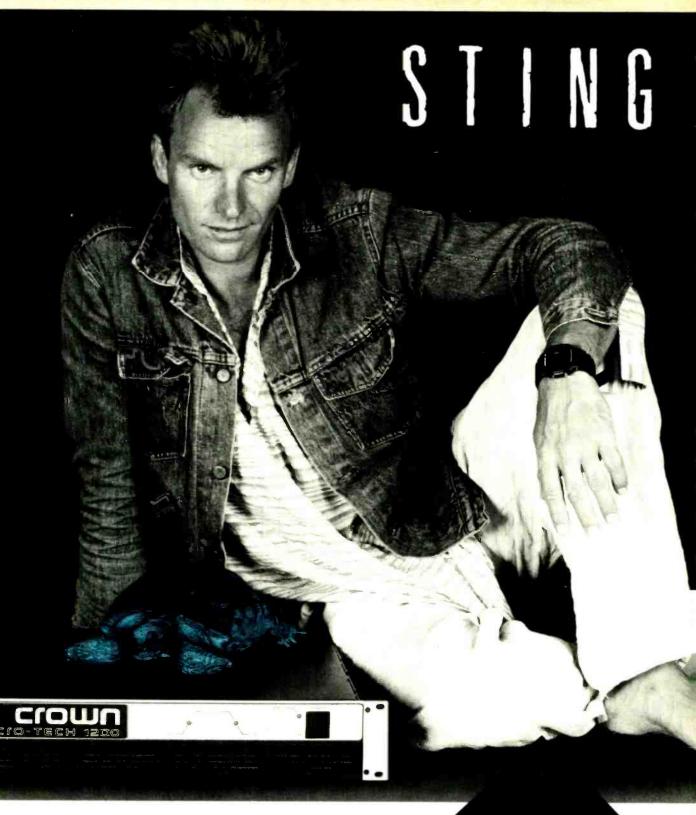
In early 1985, JBL Professional's VP of Marketing, Mark Gander, examined possibilites for a new product line that would address the market need for pre-built loudspeaker enclosures. A decision was made to offer systems that incorporated both direct-radiating and horn-loaded designs.

A new two-box, direct-radiating system developed by Stanal Sound, Ltd. (North Hollywood, CA, and Kearney, NE) and a one-box, horn-loaded enclosure assembled by Elec-

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trotec Productions (Los Angeles, CA) were deemed to have good potential for market development. OEM agreements were reached, and the concept of entire pre-built systems developed.

"Touring sound evolved over a period of years from a pile of parts used by inexperienced persons, into the sophisticated tour companies we see today using modular speaker packages," Gander explains. "JBL builds the components used in the majority of touring sound systems on the road. The fact that two of those [sound-rental] companies can come to a major manufacturer, with proprietary enclosures that can be incorporated into our product line, shows just how far the concert-sound business has come."

Gander feels that the time is right for the commercial development of entire packaged systems. "Audio equipment manufacturers are selling end-users the promise of *good* sound," he offers. "It takes a certain amount of education and expertise to take raw components, and assemble a quality sound system.
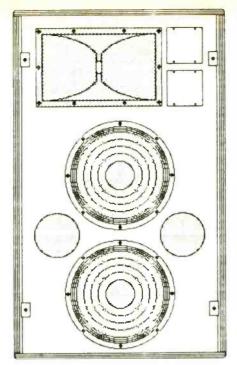
"Offering a correctly engineered system, made up of individual components that can stand alone on their own merit, is a step in the direction of *consistently* good sound for the end-users."

JBL Professional's VP of Sales, Ken Lopez, stresses that the new Concert Series systems are well-integrated. "For a matched system to be available off-the-shelf is a new concept," he concedes. "Users have the ability to go to one source for everything . . . cables, racks — all that is needed to be up and running. Our recent distribution agreement with Soundcraft mixing consoles can make the package idea even more attractive." [See News item in the February 1986 issue of *R-e/p* regarding JBL's acquisition of the sales and marketing responsibility for Soundcraft products in the U.S. — *Editor.*]

Stanal Sound Involvement

With its production wood/fiberglass shop in Kearney, NE, and considerable experience in the development of hanging sound systems and the application of fiberglass for road use, Stanal Sound was particularly well-suited for helping to develop this first commercial concert-system package.

"I have always maintained that a market existed for correctly-designed concert sound systems," advises Stanal president Stan Miller. "Many sound companies try to keep their speaker systems secret. We felt that
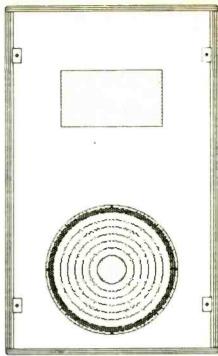
Figure 1: (left) The Model 4870 modular bi-amplified loudspeaker system houses a pair of Model 2225H 15-inch speakers and a Model 2445J driver mounted on a Model 2380 Bi-Radial horn. Figure 2: (right) The Model 4845 is a vented, direct radiator, very low-frequency loudspeaker system housing a single Model 2245H 18-inch speaker.

ours had more potential than that."

Stanal's two-box system, with trapezoidal cabinetry and integral hanging points, was incorporated into JBL's product line with very few changes. Miller considers that his experience with touring-sound systems for nearly 20 years has paid off: "We feel it is important for the primary enclosure that a system is built around to be a stand-alone device. The 4870 loudspeaker system is a full-range device with 15-inch speakers. The 4845 18-inch low-frequency box, while matching the 4870 in size, is essentially a subwoofer. Additionally, we have developed half-sized, long-throw horn boxes and two-way, downfill devices."

Loudspeaker Enclosures
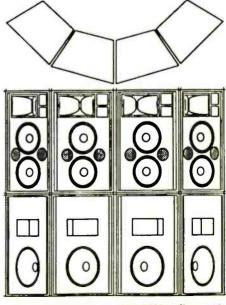
The JBL Concert Series Model 4870 is a bi-amplified speaker system comprising dual 15-inch Model 2225H loudspeakers in a ported enclosure tuned to 40 Hz. The large circular ducts incorporated into the box design allow complete freedom from vent compression over the cabinet's operating range (Figure 1).

A Model 2380 Bi-Radial™ flat-front horn loaded with a Model 2445J two-inch compression driver provides mid/high frequency coverage beyond 16 kHz. For systems requiring increased high-frequency power above 10 kHz, mounting provision and wiring are included for adding a pair of JBL Model 2404 ultra-high frequency

transducers. Cover plates for these extra mounting holes and wiring for the tweeters are standard.

The sides of each enclosure taper from front to back at a 15-degree angle. Fourteen system hanging points with Aeroquip hardware are included on each box. These points are stress-rated between 2,000 and

Figure 3: The JBL Concert Series enclosures are trapezoidal in shape, with 15-degree angled sides to enable the boxes to be assembled in tight arrays. Wide coverage can be obtained with improved low-frequency coupling.

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5,000 pounds each, depending on the angle of pull. Each Model 4870 box weighs 214 pounds, and measures 49.5 by 29.5 by 19.75 inches (H×W×D). Coverage angles are said to be 90 degrees horizontal by 40 degrees vertical, with a frequency response of 35 Hz to 20 kHz (-10 dB). The cabinets are externally finished with a dark gray, impregnated fiberglass-reinforced plastic. A black nylon protective grill completes the package.

With a 20 to 800 Hz usable frequency range, the Concert Series Model 4845 cabinet is a very low-frequency companion piece to the Model 4870. It also has identical exterior dimensions, and houses a single Model 2245H 18-inch loudspeaker in an enclosure tuned to 27 Hz (Figure 2).

The enclosures are designed to a trapezoidal pattern that allows several boxes to be assembled into tight clusters for wide-angle coverage, a configuration that also improves the coupling at low frequencies (Figure 3). The integral certified hanging points enable the quick and easy assembly of cluster arrays.

Array Assembly

The enclosures are supplied with ITT/Cannon EP-8 type connectors, one each male and female, mounted on recessed steel plates for protection (see Figure 4).

Every loudspeaker enclosure is equipped with aircraft-style pan fittings that terminate in a round-head stud. Designed by Stanal Sound, the mating hardware for rigging and hanging is available to purchasers of the new JBL cabinets. The entire system has been designed with both the structural integrity and sonic characteristics of large-scale arrays in mind.

"I have always felt that the all-in-one composite speaker boxes that were common 10 years ago were *not* the ultimate solution," Miller explains. "In assembling large arrays for arena use, there are many times when one needs to point certain components in a specific direction for optimum coverage. We have come up with a ring and stud hardware system that gives one the ability to assemble *very* sturdy clusters that, if so desired, can be permanent. Our half-sized companion enclosures for long-throw and down-throw applications give us more tools to use in quickly assembling arrays that are custom-tailored for each specific use." (See Figure 5.)

JBL's Model 4866 cabinet, which houses a pair of Model 2386 40-degree horns, is approximately one-half the size of the Model 4870 full-range enclosures. A small mid/high box

Figure 4: ITT/Cannon EP-8 speaker cable connectors are mounted on recessed steel plates.

(Model 4860) and bass box (Model 4847) are also available (Figure 6.)

According to Miller, the Concert Series enclosures were stress-tested before the box designs and construction techniques were finalized: "Whenever you suspend anything in the air above people's heads, safety is the *prime* consideration. The structural engineering is just as critical as the audio; a mechanical system is only as strong as its weakest link. We have designed this system to have *no* weak links." [See accompanying sidebar, titled Stress-Testing the Hanging System — *Editor.*]

The modular speaker boxes are easily transported on a specially-designed dolly (JBL Model 4870DL). For protection, the boxes travel face down

(Figure 7). When portable systems are transported by truck, the boxes can be stacked up to four high.

Stanal On Tour

In December 1985, Stanal took the first full-scale Concert Series arena system on tour with Neil Diamond. The brief tour played to capacity crowds in such venues as Kemper Arena (Kansas City), Riverfront Coliseum (Cincinnati), and Joe Louis Arena (Detroit).

A total of 64 Model 4873 enclosures (three-way versions of the Model 4870) were suspended in the flying array. Additionally, 32 Model 4866 long-throw packages and Model 4860 down-throw packages were carried. Subwoofer enclosures were placed at floor level.

Of specific interest with Stanal's arena array configuration for Neil Diamond were two additional clusters to supplement the main left and right hanging groups: eight Model 4873 boxes were suspended above the centerstage line, and angled downward into the forward seating sections; another separate cluster was suspended behind the stage to provide optimum sound coverage for the rear seating areas.

The two main speaker arrays were fed discrete left and right stereo program material, while the auxiliary center cluster was given a combined left/right mix. "Oftentimes in large venues, the best seats in the house have traditionally received less-than-ideal sound because of the 'hole' between the left and right stacks," Miller explains. "The center cluster helps to fill that gap, and let the closer audience areas hear a *true* stereo

Figure 5: In addition to the Model 4870 and Model 4845 enclosures, seen here are smaller long-throw and down-throw boxes that have been designed for custom-tailored arrays.

mix."

The rear cluster comprised an amount of gear equivalent to one-half of a main left or right array. Configured for 180-degree coverage, the cluster was made up of 12 Model 4873 boxes, six Model 4866 long-throw boxes and six Model 4860 down-throw boxes. To provide more than just a stereo mix, a separate mixing console, with operator Richard Albrecht, was positioned high above the backstage area, facing the rear-fill cluster. Albrecht did a remix of 16 separate subgroups fed directly from Miller's main house mix position.

"Some time ago, I noticed a hook-and-ladder fire truck going down the street," Miller recalls. "Because the truck is so long, they used a separate driver with his own steering wheel. Well, *we* like the people in the rear seats to get their money's worth. If I am out in front of the stage [mixing the show], *I* certainly can't hear what is going on back there. The separate mixing board gives us much better control for that different acoustical environment and, using subgroup feeds, still gives me a certain measure of control over the mix.*" (See Figure 8.)

For the Diamond tour, Stanal supplied an extensive stage monitoring system that rivaled the main speaker system in complexity. Although a standard central monitor mixing area existed, three of the on-stage musicians (including the band's musical director) received subgroup feeds, and were provided with auxiliary mixers. These three individuals had control over the program material content in their own stage monitor speakers. [A complete description of the stage monitor system for Neil Diamond's tour is beyond the scope of this article — *DS.*]

"It has been interesting for me to note, over the years, that the things we develop for Neil often work so well that they end up being adopted by the

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*Author's Note: To my knowledge, this is the first instance of a sound reinforcement company supplying a separate mixing console and operator for a rear speaker cluster used for large venues. Stan Miller has traditionally used Neil Diamond's show for unveiling new techniques and innovations in live-sound system technology. Over the past 15 years, these have included such new strategies as stage mixing, flying speaker systems, subwoofers, and delay-line speaker towers for large audiences. While other live-sound companies have been on similar development tracks over the years, Miller's consistent list of "firsts" with a single artist — in this instance, Neil Diamond — is perhaps without parallel — *DS.*
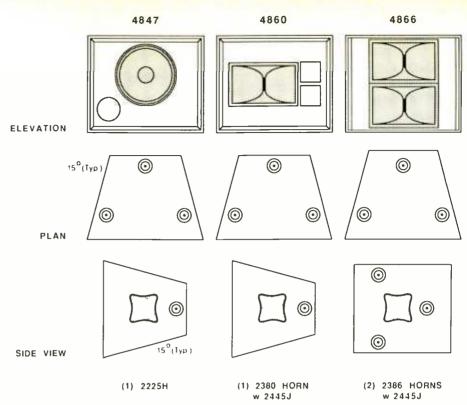
Figure 6: The Models 4866, 4860 and 4847 half-sized cabinets are fitted with the same hanging connectors as the full sized enclosures.

live-sound industry at large," Miller says. "It is quite an honor to have been able to mix the same artist's show for 18 years. When that level of trust is developed on the part of an artist and his sound engineer, those are the situations where it is possible to advance the overall 'state of the art.' Instead of having to rush things into development for last-minute tours, we are able to take a good, hard look at what direction we want to take things next. It is a mutually-beneficial situation."

Concert Series Systems

The new Concert Series system enclosures have each been given a 4800 series designation (Table 1). In addition to the series 4800 of speaker enclosures and accessories, such as cables and dollies, JBL Professional has completed formulation of the 4900 series for complete, packaged sound systems. Both direct-radiating and horn-loaded systems will be available, complete with dedicated signal processing and power amplifiers. Electronics racks and road cases also are available.

The system packages include fully-loaded enclosures, JBL/UREI electronics, and pre-assembled racks and cables. The heart of each system's drive rack is the Model 5234A electronic-frequency dividing net-

Figure 7: The JBL Concert Series enclosures travel on Model 4870DL dollies face down for protection.

David Scheirman

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● SERIES III VOICE SYSTEM

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- Velocity action.
- MIDI implementation.
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● SEQUENCERS

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- Rhythm Sequencer (RS). 16 track recorder with graphic note events.
- Composer, Arranger, Performer, Sequencer (CAPS). Up to 80 polyphonic tracks assignable to internal voices or externally through MIDI can be programmed in real time, quantized, non-real time. Extensive macro and micro editing features. Tracks can be viewed as conventional music notation.
- All sequences sync and trigger to SMPTE time code.
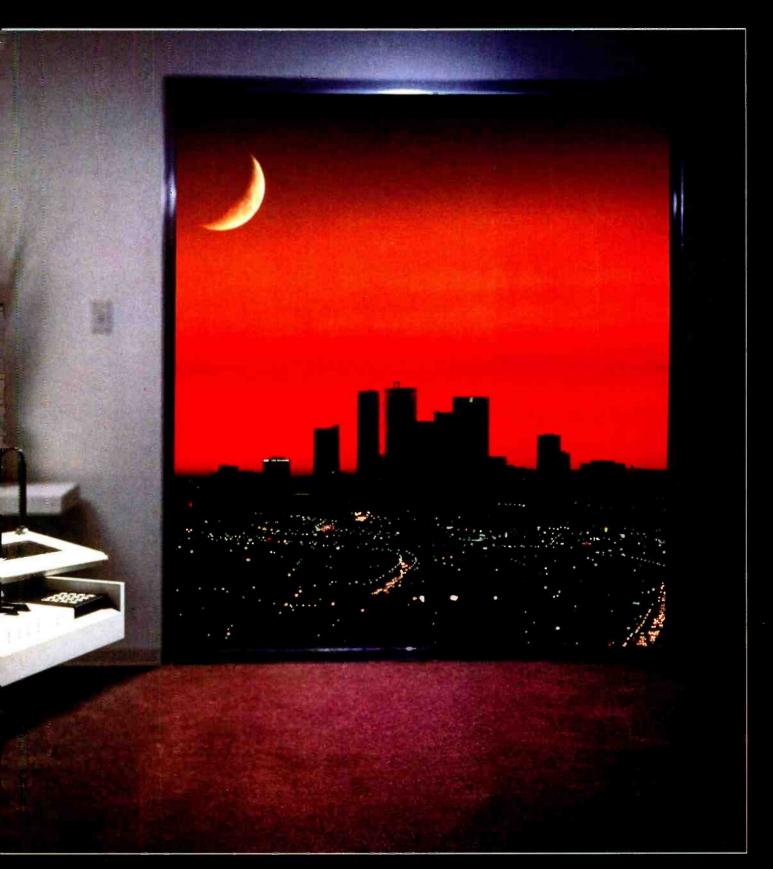
● SERIAL INTERFACES

- Dual printer ports.
- Telnet Communications software.

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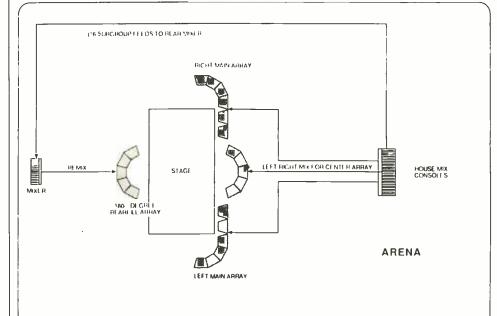
Figure 8: Typical layout of rear-fill cluster and auxiliary mixer.

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work. A variety of signal-path variations are available to accommodate different numbers and types of enclosures (Figure 9).

"Our system design philosophy is to use a high-quality class of matched components," advises JBL's Mark Gander. "Some people today are using crossovers with bandwidth shifting and independent limiting and compression for protection on the system sections. Our approach is *not* to do bandwidth shifting because of the radical changes in power response which can be incurred, and not to do limiting or compression unless every crossover section is linked to the others, to prevent wild psycho-acoustic anomalies." (The Model 5234A, used for the Concert Series, has a custom-designed, plug-in crossover card, incorporating power-response equalization and high-pass filter for low-frequency driver protection.)

Within the 4900 series systems, increasing numbers of multiple cabinets and companion electronics are available to achieve pre-specified acoustical power outputs. The 4943 system, for example, comprises all components needed to supply a maximum of 140 dB (measured at one meter, continuous program) of full-bandwidth (20 Hz to 20 kHz) audio frequency sound reinforcement from a program material input (Figure 10).

Future Directions

Stanal Sound and JBL's joint-venture development of the Concert Series systems with JBL may be significant for several reasons. The fact that a touring-sound company — an

Table 1: Component Configurations of Concert Series Systems

Model	Type	System Components
4870	Full-range	Two Model 2225Hs, and one 2380 horn on 2445J driver.
4871	Full-range	Two Model 2225Hs, one 2380 horn on 2445J driver, and two 2404Hs.
4872	Full-range	Two Model 2225Hs, and one 2385 horn on 2445J driver.
4873	Full-range	Two Model 2225Hs, one 2385 horn on 2445J driver, and two 2404Hs.
4845	Low-frequency	One Model 2245H.
4842	Low-frequency	Two Model 2245Hs.
4847	Small low-frequency	One Model 2225H.
4860	Small mid/high	One Model 2380 horn on 2245J driver.
4862	Small mid/high	One Model 2385 horn on 2445J driver.
4866	Small long-throw	Two Model 2386 horns on 2445J drivers.

Also:			
482--	Small two-way systems.	484--	Low-frequency systems.
483--	Full-range, horn-loaded systems.	486--	"High-pack" components
		487--	Full-range systems.
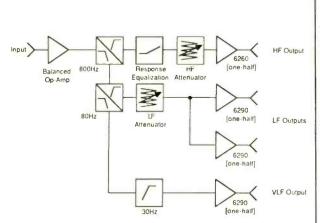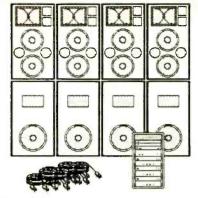
The Components of Our System

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Figure 9: (left) Block diagram of Model 4943 system electronics signal path. Figure 10: (right) Component designations of Model 4943 speaker system.

4943 Components

[4] 4870 loudspeaker systems
[4] 4845 loudspeaker systems
[4] 50 foot EP-8 loudspeaker cables
[4] 5 foot EP-8 loudspeaker cables
[1] 9943 equipment rack, including:
[1] 6260 amplifier (HF)

[3] 6290 amplifiers (2-LF, 1-VLF)
[2] 5234A dual-channel electronic crossovers with high-pass filtering, plug-in cards for 80Hz and 800Hz crossover points and power response correction.

end-user of a manufacturer's loud-speaker components — has developed a packaging concept attractive to JBL for inclusion in the latter's product line, helps to mark the maturity of the touring-sound industry. Two decades ago, sound companies looked to speaker manufacturers for enclosure concepts; 10 years ago, those same companies were hard at work developing better-sounding, more roadworthy speaker systems. Today, a new product line for mass production and marketing by a major speaker manufacturer has been developed and tested by such a touring-sound firm. The research and development cycle has gone full circle.

"Sometimes, the touring-sound companies are like our racing teams," Gander offers. "It is nearly an exact analogy to the Formula race car situation, where different drivers try out new tires and other automotive technology. We manufacture products. The touring PA's that go around the world give those products the roughest test imaginable. Having a product accepted in that industry ... *that's* our proving ground."

New lines of sound systems developed through a dialog between manufacturer and end-user will have a

STRESS-TESTING A HANGING SPEAKER SYSTEM

Hanging sound system technology for both touring and installation use can pose some very serious questions. One of these is related to hardware stress, because the methods and equipment chosen to suspend heavy loudspeaker arrays above the ground vary from venue to venue. Portable systems often rely on nylon straps and chain motor hoists, while permanent installations require more solid suspension fittings.

The temporary system places a dynamic or changing load on the rigging hardware. As the speaker array moves up and down, great force can be developed. Permanent installations are usually static, non-moving loads, although some facilities provide for the raising and lowering of arrays.

To ensure the structural integrity of Concert Series enclosures, Stanal Sound commissioned the services of an industrial testing laboratory. "It is *imperative* to know just what the limits are, and when the breaking point is going to occur," explains Mark Engebretson, an electro-acoustical consultant involved in design engineering work on the CS enclosures. "Certification of industrial hardware requires more than guesswork." Osborne Laboratories, Inc. of Santa Fe Springs, CA, was assigned the task of trying to pull apart one of the new enclosures. With 600,000 pounds of test pressure available, any weak spot in the cabinet's construction would show up.

A "sacrificial" Model 4870 system was strapped into the firm's Tinius Olsen model Universal Testing machine, comprising extremely powerful hydraulic screw-cylinders pushing in opposition. Force was gradually applied; the results are shown below.

"The device was calibrated to be accurate within 50 pounds," recalls Engebretson. "As it turned out, the box was actually *stronger* than some of the rigging hardware typically used to suspend such speaker systems in the air."

The series of three ultimate-load destruction tests on the 4870 cabinet were postponed when a nylon strap used to attach the load fasteners separated. A heavy metal clevis was then used in replacement, and the test continued.

"Eventually, the test rig just pulled the box apart," notes Engebretson. "However, it is impressive that this ultimate failure did not happen on the end-to-end test until a point well past two tons, which is beyond the capacity of the very chain motors that are typically used to support hanging speaker systems."

Such extreme stress-testing is required whenever a hardware manufacturer, or a user of hardware, needs an answer to a very important question: *How strong is it?* Many hanging sound systems use the structure of the loudspeaker cabinets themselves to support the weight of other enclosures. Verifying the structural integrity of an enclosure is one of many required steps along the path that leads to the development of safe, reliable hardware systems. □□□

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market much broader than just the touring sound industry in which it was born. "These systems are available through two different avenues of distribution: the pro-audio dealer that caters to the entertainment industry, and through qualified installation contractors," explains JBL's Ken Lopez. "Also, there will be those who bridge the gap, the tour sound accounts. It may very well be that some of the more experienced PA companies will be the model for the contracting company of tomorrow; they already have the experienced technicians in place, the fabrication abilities and the expertise in on-site troubleshooting."

When coupled with the variety of different yet compatible new enclosures available from JBL, the hanging hardware fittings for the Concert Series systems, available through Stanal Sound, point the way toward a concept that may change the process by which major venue sound system installations are bid, designed, contracted, and completed. The contractor that has the ability to quickly assemble a road-proven, modular sound system available from a reputable manufacturer in off-the-shelf form may have a distinct competitive edge.

Touring-sound company operators, installation contractors, and audio equipment manufacturers cannot help but benefit from the increased dialog and sharing of ideas that comes from such collaborative projects as the new Concert Series.

A complete, pre-tested, modular loudspeaker system available for shipping from stock, and assembled into massive arrays for large-venue use, was once only a dream. The successful completion of this project, from idea to available product, marks the coming of age for the once-maverick touring sound business, and the receptivity of an audio manufacturing industry leader to new ideas. Scribbled drawings on hotel restaurants napkins and dreams over cups of coffee have turned into engineering blueprints — from such dreams, sometimes, come future realities. If systems such as these do turn out to be compatible with market needs, Stanal Sound's years of carrying loudspeakers around from city to city would appear to have been well spent. ■■■

The mention of specifically manufactured products in this article is not to be taken as and endorsement. The detailing of such devices and description of their use has been made with reader interest and education in mind — *DS*.

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SESSION DIPLOMACY

THE TACT FACTOR

Trici Venola

Keeping Your Cool . . . While All Those Around You Are Losing Theirs

Our Saga Continues: Time is Money

by David Brody

The time: 22:30:16;04 and a couple of subframes, otherwise known as half past 10 p.m. — but it's High Noon in the mixing suite. Tension mounts. Band members are uncharacteristically quiet. Pat the Producer paces. Steve Owens, Studio Owner, stares onerously at the data acquisition LED on the transport synchronizer . . . it stays ominously dark. Vocalists fidget. Time ticks by . . .

Connectors are reconnected, patches are repatched, switches exercised, meters glanced at.

Yes, there's timecode on tape; no, the sync brain isn't seeing it. The road manager mutters . . .

Meanwhile, back at the recorder, Linda the Lone Engineer (having just mentally reviewed the salient points of the *Tact Factor,* Parts I and II) smiles and says, "Ya know, sometimes it takes a few minutes to get machines to talk to each other. I think we'll get 'em up and running in just a bit." (The girl's got class!)

Producer Pat, having now thoroughly worn a track in the control-room carpet, counters: "Yeah, well we saw it lock up a bunch of times at the studio where we printed the code: took 10 seconds the first time, and about three each time after. Shouldn't it work just as well here?"

Lone Engineer: "What kind of generator and recorder?"

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Nobody can remember.

Louie the Linndrummer: "We can call them . . . got their sticker stuck on my road case . . . 'phone number's on it."

Lone Engineer: "Let me just try one little thing first." (The Right Stuff to the max!)

She brings up the code track on the monitors. It sounds like SMPTE, but with a difference. Out loud she says: "There's something a little weird about your code." She listens a moment longer, stops tape, takes her trusty tweaker and trims back the HF playback EQ on the timecode tracks of each recorder, and tweaks up the playback level.

She hits the master play button. Steve the Studio Guy gasps as the data indicators blink on, and the slave springs to life. Kenny the Copy Room King, who's been loitering about waiting to see how this all plays out, smiles as the lock LED lights. It's now 22:41:05;16.

Good, heads-up engineering? To be sure! But also something more. Keeping the impending hysteria in check with a calm, "nothing's-out-of-the-ordinary" attitude bought the time needed to spot the problem and try some fixes. It also kept the engineer herself in a "we-*can*-make-it-work" mode, rather than getting uptight (consequently infecting everyone else's attitude to the point where the Producer called the session).

(By the way, should you ever run into this situation: certain recorders with on-board SMPTE generators throw high-frequency bit flags into the data stream as special machine commands. Rolling off the top makes the code more recognizable.) This story illustrates:

Tact Factor Axiom #4
"Any situation with both finicky machines and nervous humans has an all to easily reached threshold of Freak-Out."

Something strange and wonderful has been happening this century: We, in the Recording Arts, have been messing around with time. Specifically, we've been synthesizing perceived "realities" that alter viewer/listener subjective time — even though they may go by in real time. (And, of course, it's not merely a coincidence that this is the century of Albert Einstein and John "keep playin' those mind games" Lennon.) Engineers are charged with making the magic happen.

Unfortunately, as we who handle tasks like film/video animation and multitrack mixing know all too well, building an illusion takes tens, hundreds, even thousands of times longer than its "real runtime." Many are the artists and producers who know this in their heads, but who aren't truly prepared for the emotional tedium or the price tag. Add to this the fact that some of what we do is incomprehensible to the uninitiated — and it often looks to the outsider like we're either incompetent or wasting time.

Now, with the advent of frame-locking synchronization devices and the consequent complexity of command and control procedures for them, the problem is compounded. In this new age of super-sync, you need to command new levels of self-control and creativity. Because, as we all know:

Tact Factor Axiom #5
"The need for superior tact varies directly as the complexity of the technical task."

There are days when it just doesn't happen for you. There's a basic accursed abstinateness of mechanical objects and though they're, on the whole, getting smarter, sometimes it just takes too much time to do some things. And the worst part is that you know that somewhere, someone's building some new machine that does it all for you with two simple software commands. Anyone who's ever spent all night and several thousand keystrokes trying to get the slave reel of background vocals to lock to the proper offsets so that the producer can have her best performances of each section

THE TASCAM MS-16: FOR THOSE WHO'VE HEARD IT ALL BEFORE.

We designed our new 1" 16-track especially for the skeptics. Those who have heard all the other 16 tracks...and all the other claims. Hearing is believing, and the MS-16 delivers enough audio quality to convince the most critical ears. But that's just part of the story. The fact is, the closer you look into the MS-16, the better it gets.

The MS-16's superlative performance begins with our new micro-radii heads. They virtually eliminate "head bumps" and ensure flat frequency response. Put this together with direct-coupled amplifiers throughout, plus ultra-quiet FETs, and you get exceptional transient and low frequency response with extremely low distortion.

Unlike most tape machines, the record/sync and playback heads on the MS-16 are identical in performance, so you can make critical EQ and processing decisions on overdubs or punch-ins without having to

go back and listen a second time. You get what you want sooner and with fewer headaches.

Record/Function switches for each track allow effortless, one-button punch-ins. Input Enable allows instant talkback during rewinds. With the MS-16, you're free to concentrate on the project at hand... rather than on your tape machine.

The MS-16 takes the grief out of locking up with other audio and video machines as well. The 38-pin standard SMPTE/EBU interface affords speedy, single-cable connection with most popular synchronizers and editing systems. And the MS-16's new Omega Drive transport stands up to continual shuttling while handling tape with kid-glove kindness.

Take a closer look at the MS-16. See your TASCAM dealer for a demo, or write us for more information at 7733 Telegraph Road, Montebello, CA 90640.

THE TASCAM MS-16 SIXTEEN TRACK

TASCAM THE SCIENCE OF BRINGING ART TO LIFE.

"subdued presence on-camera"

is how a TV technical director described his favorite feature of the Bruel & Kjaer 4000 series performers' microphone. "There's no bulbous shape to clutter the scene and no bright surfaces to cause puddling or burn-in."

He might also have mentioned that the B&K 4000 series has virtually ruler-flat frequency response from 10 Hz to 40 kHz with no LF proximity effect, so you don't have to equalize it. And that the 4000 series has the least handling noise of any professional microphone in general use (64 dB equivalent SPL for 1 m/s^2).

And that the 4000 is all the microphone you need for singing, speech, instrumental or effects.

And that the 4000 is ruggedized to be equally at home in the studio or on location.

If you like to look at curves, send for our literature. If you like to hear how microphones sound in your space, call your B&K regional office for evaluation units.

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3842

For additional information circle #43

TACT FACTOR

at every one of their entrances, knows what I mean. And, of course, all the time you were dreaming of how easy it would have been to do it with that not-yet-available or too-much-money synth you've been craving. And you couldn't even tell the producer that there *was* a better way, lest she take her business to someone who's got one. Ah, the life of an engineer . . .

It is, of course, best to keep your client informed as to why a particular task will take a while to perform. But sometimes some operations are so complicated that to fully explain them would waste valuable session time, and/or make the client think that you're just showing off, thus lessening their confidence in you and the studio. Of course, "Thou shalt know thy equipment" is the first commandment of engineering ('if ya ain't got that, ya ain't got nothin'). The second is: "Thou shalt practice thy art/science humbly"; i.e., too much flash and they call you a flamer. With one eye on the clock and one hand on the checkbook, they don't have time to watch your floorshow. And speaking of clocks, welcome to:

THE DIGITAL DILEMMAS

Yes, it's that Brave New World where every sound is crisp and clean with nothing in between . . . and *so what* if the ambience occasionally gets foreshortened?? And thus, we see that there is no shortage of digital-derived sticky situations.

Integrating the Management Interface
(Protecting Your Paycheck)

Let's not kid ourselves. If a studio is recording digitally and has an appreciable amount of digital gear, then, "It's a fantastic revolutionary new technology which displays as great an auditory improvement over analog as magnetic tape did over wire." If the studio *hasn't* made the jump to digital investment, and only rents machines when the client wants digital, then "It shows promise, but still sounds very cold, sterile and artificial, and there are no reliable standard formats." So what you, Mr./Ms. Engineer, say when you're working for the studio (or walkin' around town wearing their silk tour jacket) is likely to vary directly as the equipment purchase budget of the studio; that is, *if* you want to keep your job.

It's not a fun position to be in, because you always feel like you have to compromise your honesty. You may even shy away from keeping abreast with current developments and at-

tending product comparisons for fear that you might acquire a politically unpopular opinion — and management is listening. If you quote studio policy, the client thinks you've no guts. If you say what you *really* think, there's a 50% probability of a dirty look from managment, and if you waffle on the issue, they look at each other and both laugh at you.

Well, once upon a time, some behavioral scientist put an ape in a room from which there were three ways of escaping. The ape escaped a fourth way.

The answer is that until somebody

Keeping the impending hysteria in check with a calm, "Nothing's-out-of-the-ordinary" attitude bought the time needed to spot the problem and try some fixes.

builds a viable, standard-format digital console and processing rack, it's still a partially analog process. (It is anyway, since pulse code modulation is really storing a digitized analogy of the original acoustic waveform — until you have self-replicating molecular assemblers detecting and recreating the original sound production structures in a real-time shiftable acoustic space, don't talk to me about quantum leaps or fundamental differences!) Both techniques need to co-exist (and will for some time) because certain things are easier to do one way — different things the other. Eventually, we might get to the point of chaining algorithms that simulate the warming effect of analog smear, or which add the guts and punch of head bump, or which duplicate that "constantly variable ratio" sound of tape limiting.

We're not there yet, and it may not ever be cost effective to do so. Your ability as an engineer should not depend upon what storage medium you go down to. Neither should your reputation. Don't let 'em box you in.

Having said that, there are those times when studio management wants your opinion as to what to buy. If you're on the studio's payroll, it is very tempting to say "Don't waste your money on another Super-Squash, give me a raise instead, and I'll show you how to timeshare what we've got." It is not really to your advantage for the studio you work at to be anything less than as good as it can be. If they want to make the leap to digital multitrack, it can only improve your resume to have worked with the best.

Attitude Modulation
(Them changes)

Digital recording is flawlessly unforgiving: assuming what comes out of the console is okay, you'll either get a great track, or no track at all. This calls the Tact Factor into play when you need to explain to the producer and musicians that consistency is of the utmost, and that if something too spontaneously different happens during the take, it may not live on tape. Unlike analog, you may find yourself hard up under some close tolerances and, consequently, a tad nervous. Add to this the fact that digital tape is physically more fragile (being susceptible to all kinds of demons from fingerprints to cigarette smoke), which suggests the possibility of making time-consuming back-ups, and you may find yourself having to explain to the clientle just why this digital technology is so great.

You may have to get used to a subtly different kind of pressure. Microphone and DI thermal noise will matter more now that tape hiss doesn't exist. You'll probably find yourself using higher compression ratios, but doing it less often (i.e., less compression but more true limiting). Things come back off the tape sounding more like they did going in, but there are still a different order of differences —particularly in the high-end and you've got to allow for it. You must advise the client that digital editing usually takes longer, and that setting up a digital mix for CD mastering can be both time-taxing and hair-raising. (Though much less so, on *both* counts, if you take the time to read Roger Nichols' fine article "Preparing Digital Master Tapes for Compact Disc Mastering," to be found in the April 1984 issue of *R-e/p*.)

When the client asks which digital mix-down format is the best, well of

TACT FACTOR

course it's the one the studio you work with happens to own. But make sure their record company can deal with a mastering house that can cut it, or your studio may end up having to eat the transfer time, for which you'll have to take the heat.

D I GdIiTgAiLtal D EdLeAlYay

Many engineers have a control-trip hangup: they think that only they are qualified to decide any matter which involves the tech side of the tech/artistic equation. T'ain't so, and setting up that kind of attitude at the beginning of a session can foster a delayed, multiplied negative reaction from other project participants.

More and more the lines between performer and recorder are being blurred. As digital control improves, the technology gets more friendly. Some engineers, though they may not know it, feel threatened by this. Perhaps they secretly fear their own incompetence, and need to show the world their "expertise." You can usually tell who they are — these are the guys who insist on tracking everything clean, dry, unequalized, and usually in mono.

The "We'll-add-it-later" line really no longer applies to pop music. So many sounds are delay- and ambience-dependent these days, that to take them unprocessed is to affect the musical feel — almost always for the worse. More and more musicians — keyboard players particularly — are evolving parts for which the processing forms an intimately integral part. Sometimes a guy will come into the studio and basically hand you an "effected" stereo feed. If the producer (who hired him — and *you)* doesn't have a problem with that, neither should you. On the other hand, there is a breed of musicians who are distinctly *un*comfortable with synthesized psycho-acoustic environments, and for them we must enter into:

Orchestral Maneuvers

We've been synthesizing acoustic spaces using off-axis delay lines for perhaps 15 years. It's changed the way we hear. The "big" sound of natural acoustic instruments in a hall results, in part, from players interpreting the flow of tempo slightly differently from each other, which is itself caused, in part, by their physical distance from one another. As a result, pop producers — especially nowadays — find large ensemble sections (like strings) notoriously sloppy. And, to compound the problem, these musicians tend to chatter in the rests

between their entrances.

Here's the reason: to musicians trained for the concert hall, the recording stage looks like a rehearsal studio. Having to don cans interferes with their pitch perception. The acoustic "proximity" of the rhythm or click track fools them into playing more solo than tutti (they fail to match vibrato, tend to make entrances as a mob of individuals rather than an ensemble, and so forth).

The trick is to get them up to Concert Pitch — the key is their conductor. You must follow him wherever he leads. His cue mix is more important than what you feed to the musicians; it's even more important than the control room monitor feed. Even when things are getting hairy because you're juggling many mikes and lots of limiters, spend time on the conductor's cue. Otherwise, you may end up with a flawless recording of a bad take. And you'll get blamed anyhow.

"Legit" musicians are also notorious for not being able to grasp the concept of multitrack recording. So, you need to reassure them, and sometimes spend session time proving to them that you really *can* adjust the blend so the inner parts are heard. Despite the aura that surrounds their ear-training and solfeggio, they don't necessarily hear any better than you do. Many of them are heavily into the audiophile lifestyle, so expect them to be prehyped on the "Magic of Digital." Count on them failing to understand why you would ever *want* to do a rock date if you could possibly avoid it. And the infighting for pecking

order that goes on in a string section can drive you to sell cars for a living. You may even have to erect a few unused microphones so nobody complains of being too far away to be heard! Back on the pop side of the fence, let's go:

Messin' With The MIDI

Another clock-consuming, time-related, Tact-tacky digital dilemma that can rear its ugly head in even the lowliest of four-track demo rooms is the Great Hooking Together of the Little
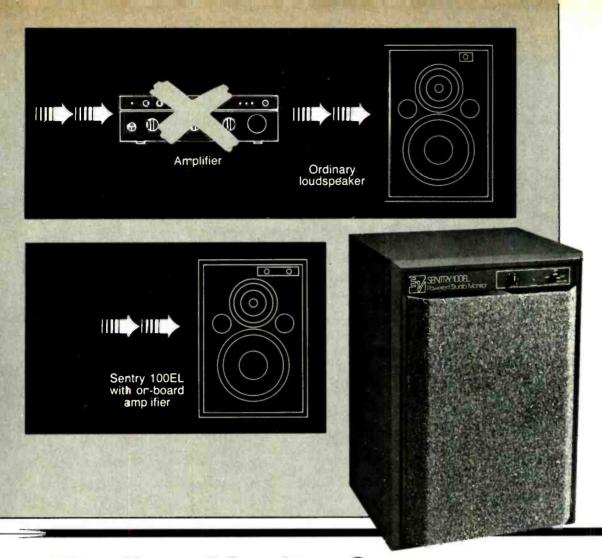

And the infighting for pecking order that goes on in a string section can drive you to sell cars for a living.

Synths. Thanks to the incorporation of the MIDI port, it's a lot easier than it used to be in the prehistoric days when everyone's idea of one volt per octave was slightly different; when sometimes the Schmidt triggered and sometimes it didn't; and when the synthesizer by the amp rack drifted ever sharper to the one by the air-conditioning duct. But, as the capabilities have increased, so has the complexity of the tasks we're asking those little circuits to perform. (And after all, they're only made of sand!)

Many of the snags arise from the demand placed on the musician (usually a keyboard player or a drummer, although these days no one is immune) to understand the abilities and limitations of the equipment — i.e., to be something of an engineer. Some of them have it all totally together. Many don't, however, and need a helping hand from you, the engineer. But no musician wants to appear incompetent in front of his/her fellow band members, or (worse) in front of a client who's hired them. So it's up to

you to "make it so" without calling attention to their lapses. Try and get some eye contact happening with the player; let 'em know you're on their side, and hope that whatever they've promised is possible.

Sometimes it *isn't* possible with the gear at hand. As mentioned in Tim Bomba's excellent article, "Recording Original Music & Sound Effects for a Videodisk Project" *(R-e·p* October 1984 issue) many MIDI sequencers have their own timebase drivers, and don't recognize foreigners from the outside world (like SMPTE). Another frequent problem area is the need to establish timebase resolution at the outset of the project. Fail to do this and you might find yourself with a control room full of eager faces demanding an effect or passage which, without an interlock box (like the ubiquitous Dr. Flick) ain't gonna fly!

Though it's not always feasible (for scheduling, financial or political reasons), the best way to avoid these things is with a pre-production meeting. Video people are used to this, audio folks less so. This is the time to look into all the pitfalls and plot a course around. Even if you get paid by the hour, it's better to have things go smoothly in the session than to waste expensive time straightening out the nonsense.

Try and get some eye contact happening with the player; let 'em know you're on their side, and hope that whatever they've promised is possible.

Once in a while, you'll come across a musician/synth operator who's just *sure* that the impossible is commonplace... and that the fact that it isn't working is your fault. You try to cool
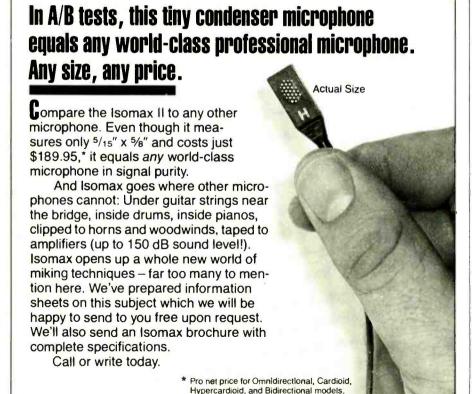
them out and explain the problem in the technical terms it deserves, but they either can't or don't want to hear you. Everyone else stands around tapping their toes impatiently while you and Sammy Synthesizer go 'round about whose pulse is failing to trigger which whatever. In this specific situation it may be better to anthropomorphize the equipment, calling it "picky" or "dumb," rather than to let things degenerate into an emotional donnybrook. It doesn't matter *who's* right, so long as you get the Best Possible Track or, in this case, any track at all.

There are usually several different ways to do anything and the listener isn't really going to care that much about how you did what you did. If worse comes to absolute worst, someone might even have to Play The Passage (Heaven forfend!), albeit at half speed and an octave lower, rather than have it trigger tracked. The human brain can be a darn good sequencer.

There are times when the long way around is the right way to go. As we move into this age of digital magic —with its higher tech and higher price tag — it's essential to try and figure out where a particular client is "coming from." Don't automatically assume that they're only interested in saving money; they may actually be more interested in spending it.

Though record companies are keeping a tighter reign on the purse strings these days, the world of corporate promotion and advertising is feeling more flush than ever. Often, you'll run into a (usually young) corporate producer who isn't making a huge salary. Thus, she or he is trying to upgrade their own resume by becoming comfortable and conversant with the upscale technologies. They will *want* you to trot out the bells and whistles; to do as much as possible with computers and other high-tech stuff, even though it may take twice as long at twice the price. If you give them only the elegantly simple solution, they'll think you're a real good Joe (or JoAnne), and proceed to go work in some other facility where their company will pay through the nose for their education — and the tape will sound about the same. This is one instance where the Best Possible Track is not the only thing at stake.

You may be called on to help your friendly local corporate producer play: "Justify the Budget." The sad fact of some corporate existences is that one's power base is defined roughly by the amount of company assets one is given to mess around with; it is often in a producer's best interests to show her boss a big bill. If she comes
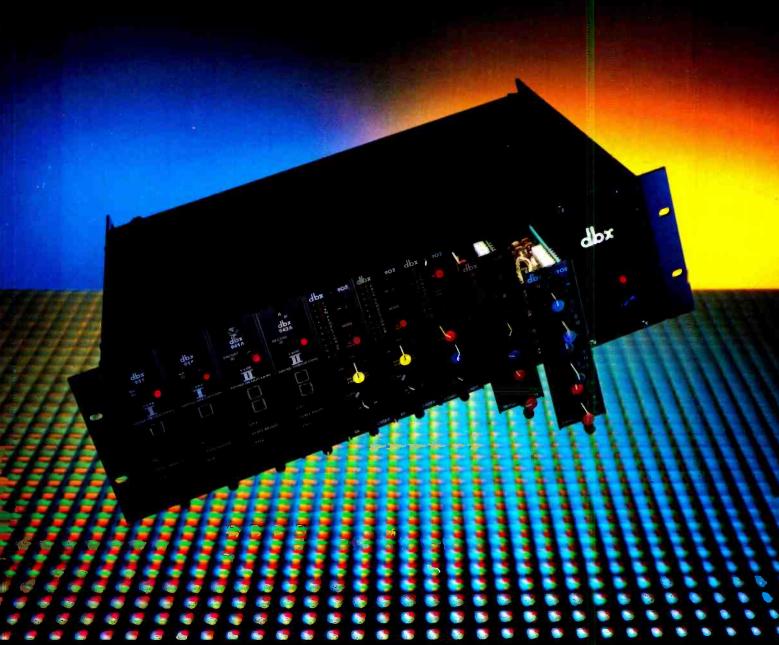
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back with an eensy-weensy invoice, the bucks may not be there next time she goes to the well.

You are basically an honest individual with a solid work ethic ('else you wouldn't be wasting your time as an engineer) so all this may leave you feeling a bit funny. Remember this mega-maxim of modern life:

*"I'd rather play facts
than pay tax!"*:
Sir Philip Fatcat
Chief Executive Officer
THE OLIGOPOLY GROUP, Int'l

In other words: What don't go to promotion gotta go to the govern-

Though the situation is improving as more broadcasters (and narrow-casters like cable networks) start transmitting in stereo, the sad fact is that audio often takes a back seat to the Pretty Pictures. You, the sound-track producer, and the score composer spend long hours, get mad at each other and generally bust your buns grabbin' those hot sounds just so some yak at Master Control can EQ and squash the living spit out of your mix. And the way they justify it when you politely call and inquire is to tell you that it sounds good emanating forth from the gloriously monophonic, six-inch Salami-skin-cone speaker (with the powerful half-ounce Alnico magnet on the back) that most view-

**Everyone else stands around tapping their toes impatiently
while you and Sammy Synthesizer go 'round about
whose pulse is failing to trigger which whatever.**

ment. All that shiny new technology that you've been drooling over at the AES and other shows is just *perfect* for satisfying corporate image. As in evolutionary physiology, Use it or Lose it.

Corporate marketing really is a different animal from the entertainment business, and we must make an extra special effort to understand and be understood. They generally walk the straight and narrow; we often operate near the fringe. They're bound to wonder what we're really into anyway, dealing daily as we do with gates and keys and harnesses (not to mention razor blades)! To the Big Business mind, audio content is *vastly* more important than audio quality. *In a related story:*

ers still have as their only link with audio reality. Well, it sounded good in the studio, you think, as you drown your sorrows in the warm wash of somebody else's magnificently digitized orchestral CD.

There are ways around bits of this: some technical (stereo delays on the order of 25 milliseconds do not fare well through the A+B/A–B matrix process, and pitch-shifted splits on guitars are notorious for collapsing entirely when folded to mono), and some tactful. From the outset of the project, stress to your client that the final product will be an accompaniment to a visual broadcast and that, therefore, things might need to be more open than otherwise. This is especially true if you're not going to be

mixing to picture. Of course, reference everything on Cruddyphonic monitors (but then you do that anyway, right?) to make sure nobody gets married to subtlety. Then, when all is said and dubbed, label your final: Equalized for Air. They may change it around anyway — but perhaps they'll be lazy and won't. Hang in there; the Recording Arts interface better with each passing nanosecond.

Future Factors

Some of us were doing automated mixes 10 years ago. More industries are now implementing computers and automation. So are more consumers, and a common language is developing. With this increasing computer literacy comes a common understanding about the tricks and techniques of human/machine interaction. There will still be many situations that'll make us feel like we want to dive into one of the apertures in the Sonex™ and disappear, but more and more performers, producers and the general public are getting hip to the wiley ways of devices. That can only make recording the Best Possible Track easier in the long run. (For example, the modern dance record, as an artistic form, was totally invented by its own technology.)
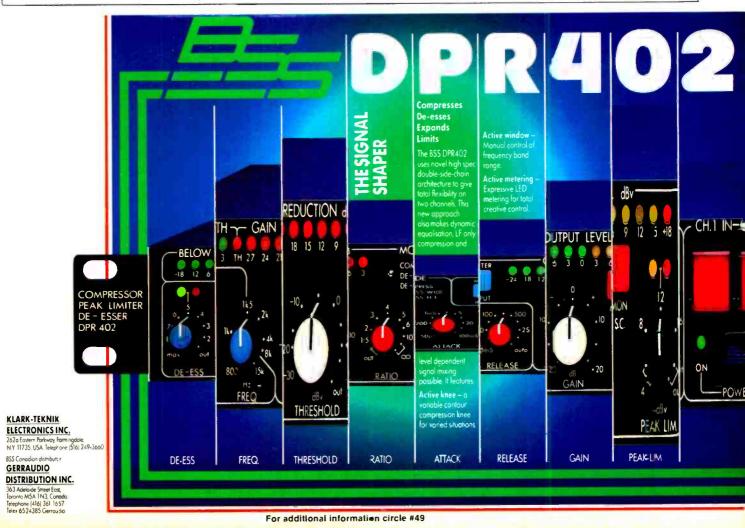
So time marches on, ticks by or
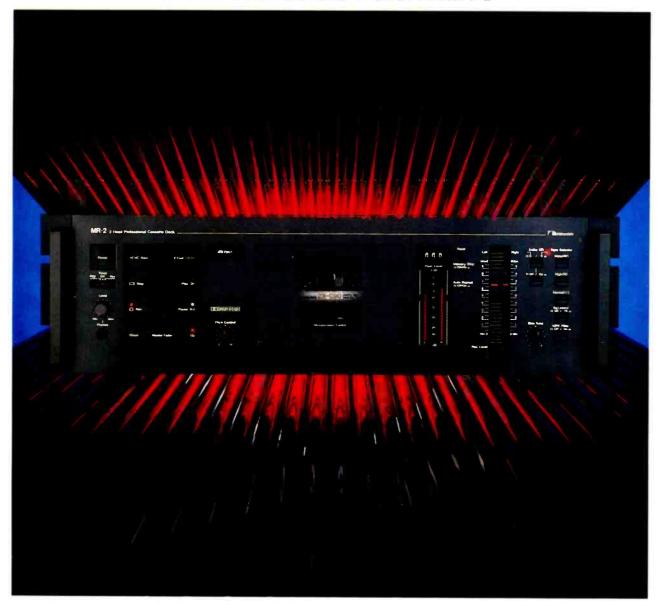
Any situation with both finicky machines and nervous humans has an all too easily reached threshold of freak out.

crystally resonates. Our machines and our human friends are becoming more interactive — with themselves and each other. As we move out onto the digital frontier, perhaps we Lone Engineers won't be so lonely. After all, to be truly user friendly, a device must incorporate into its design, modes of operation that at least simulate The Tact Factor. ■■■

Northeast:

☐ **MILLBROOK SOUND** (Millbrook, New York) has installed **Tannoy SFM** double 15-inch dual concentric monitors with **Sync-source** crossovers, coupled to a **Tannoy SR840** class A MOSFET power amplifier. Also added were **Tannoy NFM-8** close-field monitors; a **Klark Teknik DN780** digital reverb/processor; two **Lexicon PCM-60** effects units; an **Em-u Systems Emulator II, Roland Super Jupiter** and **JXP8** synthesizers; an **Oberheim DX MIDI Stretch** drum computer; an **Aphex Compellor**; and a **Bechstein** grand piano. In addition, the studio now offers accommodations and transportation services. Plans are also being made to offer in-house engineering and production seminars. *PO Box 317-Route 44, Millbrook, NY 12545. (914) 677-3744.*

☐ **WCBR** (Waltham, Massachusetts) has installed a **Neve Model 5432** production console for use during broadcast of the New York Philharmonic Orchestra. This latest purchase is the second Neve console for the facility. *750 South Street, Waltham, MA 02154. (617) 893-7080.*

☐ **CENTURY III TELEPRODUCTIONS** (Boston, Massachusetts), a video production and post-production facility, has named **April Baldwin** as account executive for its audio division. Baldwin, who previously worked for Kevin Tracey Productions, Inc., has a Bachelor of Arts degree from Tufts University. Also **Susan Roche** has joined the company as engineering technician; she received her BA in Broadcast Production from the University of South Florida. *651 Beacon Srteet, Kenmore Square, Boston, MA (617) 267-6400.*

☐ **NBC NETWORK** (New York City) has taken delivery of three **Neve** consoles. The first is a 12-input **Model 5452** broadcast console for use in the network's remote video truck; the second is a 12-input **Model 5465** for NBC Television News; and the third is a 16-input **Model 5455** console for use during equipment analysis. *New York, NY.*

☐ **VERBATIM RECORDING** (River Edge, NJ) is a new synthesizer-based, pre-production and production facility that boasts a **Soundcraft Series 200B** console, **Otari MX-5050 MkIII** eight-track on half-inch machine, with monitoring powered by **Crown** amplifiers. All sequencing is handled via a **Apple IIe** personal computer and a **Roland SBX-80** Sync Box. Outboard gear and instruments include a **Roland MSQ-700 Midi Bass, Lexicon Models PCM-41** and **PCM-42** digital delays, **Symetrix Model 501** noise gates, and a **Yamaha RX-11** drum machine. *PO Box 372, River Edge NJ 07661 (201) 262-8255.*

Southeast:

☐ **OMEGA RECORDING STUDIOS** (Rockville, Maryland) is a new 12,000 square-foot, three-studio complex, that recently merged its two separate facilities into a single location. Owner **Bob Yesbeck,** says the reason for the recent merger was because "it was difficult to run two studios at two locations. For cost-effective and convenience reasons, we are now in one place. Plus we can now 'mix and match' tape machines and signal processing equipment among all our control rooms." The move also brought the opening of a Studio C — a 25- by 25-foot studio working into a 25- by 25-foot control room, the latter of housing an **Auditronics** production board, **Studer A80 MkIV** 24-track, two **A810,** plus three **B67** two-tracks, an **Ampex ATR800** four-track and **AG-440** eight-track. The radio/TV post-production room also boasts a new synthesizer array consisting of an **E-mu Systems Emulator II, Yamaha DX-7, Korg DW800, Roland MIDI Bass,** and **IBM** and **Macintosh** personal computers. The **Sound Ideas** Compact Disc sound-effects library is also available, in addition to a 48-line tie line linking control rooms A and B to Studio C. *5609 Fishers Lane, Rockville, MD 20852. (301) 946-4686.*

OMEGA — forms three-studio complex

☐ **NEW AGE SIGHT & SOUND** (Atlanta, Georgia), reportedly the city's first digital recording facility, has hired as staff engineers **Mitchell Dorf** (a recent graduate from the University of Miami), and **Jason Bonnette** (a graduate from the Music Business Institute). In addition, entertainer/promoter **Jack English** is now representing the studio in the sales and marketing department. *120 Interstate North Parkway East, Suite 164, Atlanta, GA 30339. (404) 956-7956.*

☐ **ALPHA AUDIO** (Richmond, Virginia) has re-opened Studio II to specialize in sound effects for commercial production. The refurbished room boasts a 20-in/eight-out custom **Quad-Eight/Westrex Model 248** production console, **Otari MTR-10** and **MX-5050** two-tracks, two **Yahama REV-7** digital reverbs, **Orban Model 424-A** stereo limiters, and **Valley People Kepex** noise gates. The studio was designed by **Alpha Audio's Professional Equipment Division,** with acoustic design contributed by the company's **Acoustics Division.** *2049 West Broad Street, Richmond VA. 23220-2075. (804) 358-3852.*

☐ **B&B PRODUCTIONS** (High Point, North Carolina) is a new 16-track facility geared for album and music production. Featured is a 24-input **Soundcraft Series 600** console linked to an **Otari MX-70** one-inch 16-track, and an **MX-5050 Mk III** two-track. Signal processing gear includes an **Eventide Model H949 Harmonizer; Lexicon PCM-42** digital delay and **PCM-60** digital reverb; **dbx Model 160X** compressor/limiter; **Symetrix Model 511** single-ended noise-reduction system, and combination **Model 522** compressor, limiter, expander, gate, and ducker; and **US Audio Gatex** noise gates. Monitoring is provided by **JBL 4401**s and **Polk Audio 10B**s. Microphones are from **Neumann, AKG, Electro-Voice, Beyer,** and **Shure.** *Route 2 Box 653, High Point, NC 27260. (919) 769-9171.*

South Central:

☐ **THE ROCK STUDIO** (Norman, Oklahoma), a division of **Rock Studio Supply,** has taken delivery of a new 32-input, 24-subgroup **ACES** console, plus a 24-track from the same manufacturer. *PO Box 5997, Norman, OK 73070 (405) 329-8431.*

☐ **JANUARY SOUND** (Dallas, Texas) has completed its **Sierra Audio**-design/renovation of Studio A. To complement the new acoustics, the facility has purchased a **Sony JH-536-D** automated console; a **3M M81** digital 32-track, plus a pair of four-tracks with full digital editing capabilities, a **Sony/MCI JH-24** multitrack and **JH-110B** half-inch two-track. Outboard gear comprises **Klark Teknik DN780** and **ART DR1** digital reverbs; and microphones by **Neumann** and **AKG.** To ensure compatibility between the two studios, Studio B also houses a new **Sony JH-24** and **Model JH-536** console. *3341 Towerwood Drive, Dallas, TX 75234. (214) 243-3735.*

Midwest:

☐ **METRO STUDIOS** (Minneapolis) has completed a quarter-million dollar upgrade with the acquisition of the following

equipment: a 30-input **Trident Series 80B** console with **Sound Workshop ARMS** automation; an Otari MTR-90 MkIII 24-track with autolocator; **Sony JH-110B** and **Scully 280B** two-track machines. Outboard gear acquisitions comprise an **AMS RMX-16** digital reverb, **Eventide SP2016** effects processor, EMT 240 Goldfoil reverb, and a **Studio Technologies Ecoplate II** reverb; **AMS RMX 15-80S, Lexicon PCM-42, PCM-41,** and **Roland SDE-1000** digital delay processors; four API 55A equalizers; **two Aphex EQF-2** equalizers; an **Aphex Compellor;** and **UREI LN1178** compressor/limiters; **Valley People Kepex** and **Gain Brain IIs;** two **Valley People 430s;** and an **Eventide H-910** Harmonizer. Monitoring is provided by JBL Model 4430s, Model 4411s, and Model 4311s; Yamaha NS-10Ms; and **Auratone 5-Cs**. Microphones are by **Neumann, AKG, Electro-Voice,** and **Shure.** Finally, the facility has named **Tom Tucker** as partner and chief engineer. *200 Third Avenue North, Minneapolis, MN 55401. (612) 338-3833.*

☐ **GNOME SOUND** (Detroit) has added a **Yamaha REV-7** digital reverb/effects processor and an **Aphex Compellor** complimiter to its outboard equipment rack. In addition, **Kevin Knapp** has been added to the facility's staff personnel as director of media relations. *PO Box 28284 Detroit, MI 48228. (313) 835-0169.*

☐ **MINNESOTA PUBLIC RADIO** (St. Paul) has added two **Neve** consoles: a 36-channel **V-Series** broadcast desk for the facility's Studio M, and intended for local-area jingles and audio-for-video post production; and a 36-channel **Model 5106** for the newly renovated **World Theater**. *45 East 8th Street, St. Paul, MN 55101 (612) 221-1500.*

Southern California:

☐ **ENCORE STUDIOS** (Burbank), formerly **Kendun Recorders,** has completed the following studio renovations: a new hardwood floor, replacing the absorptive, wall-to-wall carpeting; plus a fully motorized system of acoustic louvers for the walls and ceiling, remote controllable from the control room. Acoustic upgrades were carried out by **Lakeside Associates,** under the supervision of **Carl Sanchar.** The control room, orginally designed by **Tom Hidley,** has been left virtually intact, including the the original **Sierra/Hidley T-8** bi-amped monitoring system. "We can now offer the best of both worlds," says studio **Billy James.** "This has always been one of the most renowned control rooms in the country. With the new, live-sounding tracking room, we can accommodate a wide variety of recording dates as well." These acoustic upgrades are coupled with the facility's existing **Solid State Logic SSL4000E** mixing console with **Total Recall,** linked to two **Studer A-800** 24-tracks and two **A-800** two-tracks. Complementry outboard gear includes **Lexicon 224XL,** an AMS digital delay, **a dbx model 902, Drawmer noise gates, and two EMT 140** tube plates. *Burbank.*

☐ **MASTER CONTROL** (Burbank) has added a pair of **Drawmer** noise gates and a **Yamaha REV-7** digital reverb. It is also reported that the studio plans to install a 52-input **Solid State Logic SSL 4000E** console with **Studio Computer** and **Total Recall** automation. *3401 Burbank Blvd., Burbank CA 91505 (818) 842-0800.*

☐ **LARRABEE SOUND** (Los Angeles) has opened its new Studio B, equipped with a 56-input **Solid State Logic SSL4000E** console and **Total Recall** computer automation (this recent acquisition is the second SSL board in use at the facility). Also available are two **Studer 800** 24-tracks; two **820A** two-tracks; and a **Mitsubishi X-80** Prodigital two-track. Outboard gear comprises two **AMS DMX 4.8,** two **DMX 6.4,** and three **RMX** digital reverb/processor; two **Lexicon PCM-70s,** four **PCM-42s,** a **Model 200** and **Model 224XL** reverb/processors; an **EMT 252** plate reverb; two **Yamaha REV-1s;** and two **TimeLine Lynx** timecode synchronzing modules. *8811 Santa Monica Blvd., Los Angeles CA 90069. (213) 657-6750.*

☐ **CHARTMAKER** (Los Angeles) is the name of a recently opened, 24-track private-use studio owned by producer **David Foster.** Designed and built by **Lakeside Associates,** the 500 square-foot control room boasts a 40-input **Solid State Logic SL4000E** console (pictured here with Foster's regular session engineer **Humberto Gatica**); a **Studer A800** 24-track and two **A80** two-track; four **Aiwa 660** cassette decks; and a **Sony** CD player. Outboard equipment includes an **EMT 250** reverb; **Lexicon 224XL** digital reverb; **AMS DMX** digital reverb and delay units; **Drawmer** noise gates; **dbx** limiters; **Teletronix LA-2A** limiters; and an **Eventide H949 Harmonizer.** Monitors comprise two pairs of **Yamaha NS-10Ms,** and a three-way JBL system. In addition, built-in direct boxes and 32 MIDI tie lines are accessible throughout the studio, utilizing a **J.L. Cooper 16/20** MIDI routing box for patching different synthesizers. Foster's new studio is said to have been designed mainly for album production, plus audio-for-video, and -film mixing and scoring. *Los Angeles.*

Northern California:

☐ **PRARIE SUN RECORDING** (Cotati) has acquired an **E-mu Systems Emulator II** digital synthesizer, and an **Apple Macintosh** personal computer with external 20-Mbyte hard-disk drive, running the **DigiDesign Sound Designer** MIDI-programmable software. *PO Box 7084 Cotati, CA 94928 (707) 795-7011.*

☐ **SRO STUDIOS** (San Francisco) is a joint venture between two pro-audio companies: **The Sound Service** and **Studio C.** The new three-room complex will focus on audio-for-video and -film post production. Studio 1 will perform transfers of tape-based audio to any other tape format, utilizing a 28-input **Neotek Series 2** console, **Ampex MM-1100** 16-track machine with **Dolby A** noise reduction, **Audio Kinetics Q.Lock** timecode synchronizer, **Sony JH-110** two- and four-track machines, and a **JH-110** one-inch video layback machine. Studio 2 is a film mixing theater, equipped with a 32-input **Auditronics Model 501** console. Still under construction is Studio 3, an Expansion Room, and which will house a 24-input **Auditronics** board with corresponding mutitracks

LARRABEE — second SSL acquisition

CHARTMAKER — Foster's 24-track facility

PIONEER — new ADC UA8000 console

and sychronizing systems. In addition, the facility is equipped with two dedicated audio-format transfer rooms, and a tape-based sound effects library. *1338 Mission Street, San Francisco, CA 91403. (415) 863-0400.*

□ **CITY SOUND RECORDING** (San Francisco) is the new name for the former **AKA City Sound Studios**. The studio recently purchased two **Lang TEQIIs**, two **Pultec EQs**, a second **AKG C12** microphone, an **Otari MTR-10** two-track machine, an **Audio+Design/Calrec Panscan**, and a **Fairchild Model 660** tube limiter. *245 Hyde Street, 2nd Floor, San Francisco CA. (415) 474-0377.*

Australia:

□ **POWERPLANT RECORDINGS** (Melborne), formerly an eight-track studio, has installed a **Trident Series 80B** console and a **Studer A80 MkIV** 24-track machine with 20-point autolocator. Other acquisitions include a **Studer A810** two-track, two **B77s**, **Drawmer** noise gates and compressor limiters, an **Orban 424A**, a **Roland SDE-3000**, **Linear Transfer** power amplifiers, and various microphones from **Sanken, Sennheiser, and Shure**. *2-24 Little Cardigan Street, Carlton, Australia 3053. (03) 348-1300.*

Canada:

□ **COMFORT SOUND** (Canada) has installed an **Aphex Stereo Compellor** compressor/limiter and added an **AKG BX20** reverb to its studios. In addition, the facility's owners are currently evaluating a new control room monitoring system. *26 Soho Street, Suite 390, Toronto Ontario M5T 1Z7. (416) 593-7992.*

England:

□ **ABBEY ROAD STUDIOS** (London) is reported to be taking delivery of a new 36-input **Quad Eight/Westrex** console for Studio 3, featuring low-noise mike amps and mixing circuitry for "full compatibility" with digital recorders, in addition to the instalation of VCA subgroups and parametric equalizers. *London.*

□ **ROOSTER STUDIOS** (London), has taken delivery of a **Mitsubishi X-850** Prodigital 32-track machine, which will be linked to the facility's existing **Harrison MR3** console, cequipped with **Audio Kinetics MasterMix** disk-based automation. With the X-850 acquisition, studio owner **Nick Sykes** says that he is hoping to phase out all analog-based sessions within six months. *Shepherds Bush, West London. (01) 602-2881.*

Japan:

□ **PIONEER LASERVISION** (Japan) has installed a 48-input **Audio+Design/Calrec UA8000** mixing console (pictured on previous page) with 48 channels of **Audio Kinetics Mastermix** VCA automation. Outboard gear comprises a **Cipher Digitl Softouch** synchronizing system and **Lexicon 224XL** digital reverb. Monitors are **JBL/UREI Time Aligns**. *Japan.*

STUDIO PROFILE: Sellar Sound Re-Opens As 24-Track Facility
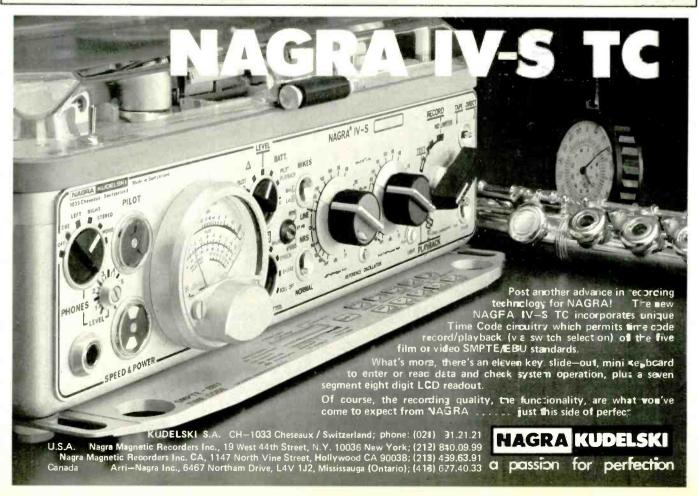
☐ **SELLAR SOUND** (Sterling Heights, Michigan) has re-located to a residential area, and re-opened as an album-oriented studio tailored for local-area Detroit bands. Owner **Gary Spaniola** says that the relocation is a direct result of his former studio control room being too small to comfortably house a new 44-input custom-designed **Neotek Elite** console. Currently, the new control room measures 22 by
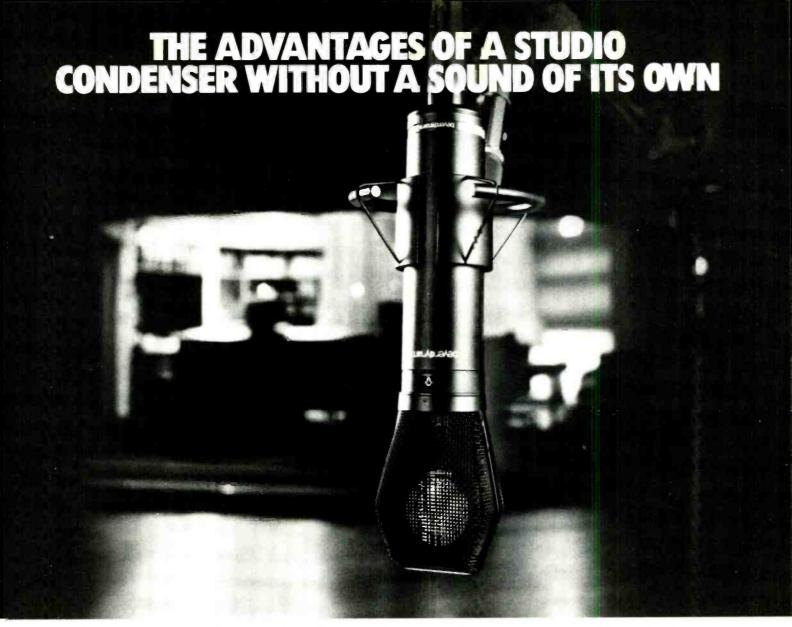
16 feet, and the studio 25 by 27 feet, with two guitar and drum isolation booths "that act as live chambers," says Spaniola. Housed in the control room **Otari MTR 90 MkII** 24-track, **MTR 10** two-track, and **MTR 10** half-inch machines. Outboard gear includes a **Lexicon 224XL** with **LARC** controller, **PCM-94, PCM-42,** and a new **PCM-70** digital reverb; **Eventide SP2016** and **H969** processors; five **Valley People Gain Brain II**s, five **Kepex II** noise gates and **Maxi Q**s; **Pultec** tube equalizers; **Telectronics** tube limiters; **Deltalab DL2** and **DL4** digital delays; **UREI LA4A** compressor/limiters; and **Studio Technologies Ecoplate II**; microphones from **Neumann, AKG, Sennheiser** and **Shure**; and a 512-Kbyte **Macintosh** personal computer with external hard-disk drive, running

SELLAR SOUND — Neotek Elite upgrade

DigiDesign's **Sound Designer** software for the **E-mu Systems Emulator II** digital sampling keyboard. Monitoring is supplied by **UREI 813B**s, and for close-field reference, **Audio Reference AR18** speakers mounted on the console. According to Spaniola, the facility's equipment collection has grown steadily since he first opened his studio. "Because I'm a musician, I began as a four-track studio for my own use. When I found that I wanted more gear, I had to open my doors to clients to help pay off my existing equipment, and to allow me to buy more technical gear." *5014 Peeksville Drive, Sterling Heights, MI 48077 (313) 264-7500.*

AUDIO PRODUCTION FOR BROADCAST

DIGITAL RECORDING AND RANDOM-ACCESS EDITING TECHNIQUES

A Spotlight of Modern Videofilm's New Facility and a Conversation with post-production mixing engineer Jerry Clemens

by Ralph Jones

In a move that has stimulated considerable discussion within the West Coast post-production community, dubbing engineer Jerry Clemens recently left his post at Compact Video, Burbank, CA, to join the staff of Modern Videofilm, Hollywood. A well-respected professional with over 20 years of experience in audio sweetening for TV and video, Clemens has garnered some 20 Emmy nominations and two awards (both for music specials). In a succession of earlier moves — first to Sunwest Studios, then to Glenn Glenn, Hollywood; and

finally, to Compact Video — he has followed a career pattern that bears directly on his current situation and explains, in part, the reasons for the current speculation.

In the cases of both Sunwest and Compact, Clemens established new audio post-production facilities that subsequently became highly successful. In each of his previous positions, he has been on the cutting edge of developments in electronic post-production techniques. And, in his current position at Modern Videofilm, Clemens' job is — you guessed it — to

supervise construction of a new, state-of-the-art audio studio designed and equipped specifically for electronic post-production.

To learn more about the current renovations at Modern Videofilm, *R-e/p* sought out Jerry Clemens, who was hard at work supervising construction of the new facility. Our discussion ranged over a variety of topics, from the background of Modern Videofilm and specifics of the planned sweetening room, to broader issues affecting the near future of electronic post-production techniques.

Facility Planning

Located on Sunset Boulevard in the heart of Hollywood, Modern Videofilm enjoys both a convenient location, and a measure of respect in the film and video communities. Owner Moshe Barkat founded the company in February, 1980, to provide Telecine services and audio dubbing for syndicated programs, later adding video editing services. Today, a substantial portion of Modern's business is multi-language soundtrack conforming — an involvement that has grown, in part, from the company's work with film-to-tape mastering for the videocassette market.

Modern Videofilm has established a reputation for high-quality videocassette transfers and foreign-language releases, chiefly because of an early commitment to digital audio. In the case of most features that Modern handles, audio is transferred from the original 35mm mag film dubbing elements, directly to a Sony PCM-3324 digital multitrack interlocked via timecode with picture. The various foreign-language dialog units are then laid onto separate tracks, allowing the dubbing mixer to provide any desired final mix.

"The use of digital recorders really preserves the quality of the soundtrack," Clemens offers. "We make the 24-track digital transfers, put the audio into format, and then usually send a one-inch PCM-1610 or 1630-format digital master of the final mix to the house that makes the videocassette copies. So, the person who buys a Beta or VHS HiFi cassette is only *two* digital generations away from the original mag. There's no way for the consumer to get any closer to what it sounded like on the dubbing stage, other than actually being there for the dub!"

Moshe Barkat's recent decision to hire Clemens represents an effort to build on Modern Videofilm's success in film-to-tape mastering, and expand the company by entering the audio post-production and sweetening mar-

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ket. In the process, both Barkat and Clemens have agreed upon maintaining a commitment to the audio quality for which Modern is known, and to furthering the use of digital audio technology in audio post production.

Current Renovation and Equipment Selection

According to Clemens, Modern's long-range plans call for two sweetening rooms and, eventually, a film re-recording room. In the first phase of the renovation (currently in progress, with a completion deadline in April), the company is constructing a single sweetening room to augment an existing audio facility.

"In order to complete our plans," Clemens relates, "we've got to get out of this building in the next year or so: there's just not enough space. So, right now, we're building a room that's not as lavish cosmetically as we would like, though it uses state-of-the-art equipment. And it's being constructed in such a manner that we can literally unplug the hardware, roll up the cables, and take it to the new building."
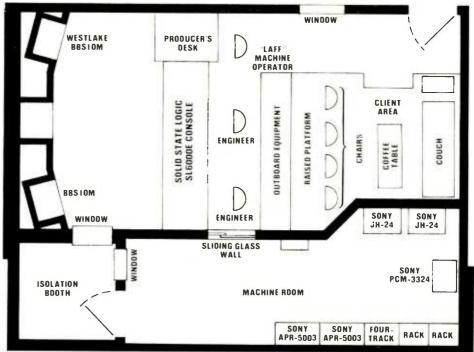
The room currently under construction will measure 25 by 15 feet, and will incorporate an adjacent machine room and vocal booth for voice-overs or simple dialog replacement. Architect Jeff Cooper has been retained to provide acoustical design.

The initial complement of analog machines will comprise a pair of Sony JH-14 24-tracks, two Sony APR-5003 two-tracks, and a four-track. While the exact model of four-track was undecided at press time, Clemens predicts that he will probably purchase an Otari MTR-10, "because we still need 14-inch reel capacity. Some clients want to do M&Es [music and effects submixes] on four-track at 15 ips [with timecode], and you need to be able to do an hour load-down."

Additionally, the company's Sony PCM-3324 digital multitrack will be available for sweetening as required. Sound effects will be handled from four ITC NAB broadcast cart machines, supplemented by a Nikko NCD-600 Compact Disc player and a random-access library on CD.

Centerpiece of the new facility will be a fully-automated Solid State Logic SL-6000 console, modified to provide an eight-stripe mix (rather than the more common six-stripe of stereo music, effects and dialog).

"With this modification, the console will allow us to provide clients with foreign-language audio dubs broken out in separate music, effects, dialog and audience reaction tracks

Floor Plan and Equipment Layout of Modern Videofilm's new post-production facility. Overall room dimensions are 25 by 15 feet, including a separate machine room and vocal booth.

— all in stereo," Clemens explains. "With the six-stripe format, audience reaction cannot be recorded as a separate track. But many of our clients want the option of handling that separately."

Clemens has specified a 48-input console mainframe configuration with dual-line inputs, providing an effective total of 96 available inputs. "I requested only four mike inputs," he relates, "because we just don't do much of that kind of recording."

The facility will feature Westlake BBSM10 monitors and, in Clemen's words, "every kind of outboard gear imaginable," including: an Aphex

— Jerry Clemens —

Compeller, Dominator, and Aural Exciter; JBL/UREI LA/4A, Models 1176, 1178, and 565 dip filter; Lexicon Super Prime Time 97, Models 224, and 200; Yamaha REV-1; four Valley People Kepex noise gates, and four Gain Brains. The current plans call for a synchronizing system based on Time-Line Lynx modules.

"We feel that, down the road, Lynx will work the best for us," Clemens offers. "We've tested it, and it seems to interface very easily with most any machine." Such interface capability is important to Clemens, since one of his criteria for the new facility will be the ability to lock *any* combination of audio transports with picture from ¾- or one-inch videotape. Clemens maintains that the new sweetening facility will be able to accommodate NTSC- or PAL-format video "at the push of a button."

In addition to building the new sweetening room, Modern Videofilm is also renovating a smaller, existing audio studio. "The existing studio will be used mainly to handle laydowns, laybacks, film conforming and some simpler mixing jobs," Clemens relates. "That room is being refitted, repackaged and cleaned up a bit."

Electronic Post-Production

It seems clear from the design of their new sweetening room that both Clemens and Modern are firmly committed to electronic post-production techniques — an impression that Clemens confirms. "We're really trying to bring videotape and film re-recording closer together," he elaborates. "Every

one in town is working to merge these two industries. In weekly episodic television, for example, it's very common to shoot in film, then transfer to videotape and edit. It would seem to make sense, then, to finish the sound-track from videotape as well."

As everyone involved in audio post-production knows, the transition to electronic post is not as simple as it might appear. While synchronization standards no longer stand in the way, other concerns now have surfaced — and there are still some arguments in favor of working in mag film. Clemens is acutely aware of such concerns, and of the need to address them. "Quite frankly," he confides, "some things are *still* easier on film. Consider the problem of editing dialog, for instance. On magnetic film, you have one sprocket hole resolution [about 10 milliseconds]. But electronic editing is a different story: when you put an audio machine in record and observe the bias risetime, you're looking at anywhere from two to five

frames [65 to 165 milliseconds] resolution — and that's *unacceptable* for dialog editing. Furthermore, with 35mm mag, if you get a little snivet on the end of a word you can scrape it off with a razor blade, or a little bit of acetone, and it's gone. On videotape, you can't do that!"

The notoriously close deadlines of television production also pose difficulties for electronic technologies, when compared with traditional magnetic film, as Clemens explains: "Let's say you've got a show with a major deadline. Typically, an electronic post-production facility may have only one or two prelay rooms. That being the case, the only way you can rush that show is to work round the clock.

"But in film, if it's a heavy effects show, you just get more Movieolas and more editors: in effect, you spread the work horizontally. At this point, you can't do that on videotape because prelay systems are expensive to build. By the time you assemble a multi-track, a couple of quarter-inch machines, a synchronizer and a ¼-inch videotape machine, you've spent close

to $100,000 for the system. But Movieolas are cheap to rent — I think they're about $85 a week!

"So, for people who want to finish a film show on videotape, we've got to find a faster way, or it's just never going to get there! And yet, it seems ridiculous that we finish on mag, with six or seven people building sound effects, when image editing technologies have come so far. We've got to get the sound to catch up."

Clemens is counting on digital audio recording and editing systems to bridge the gap. "Ultimately, digital technology can give us the ability to edit dialog, effects and music, and move them around — a lot more easily than we can on tape." he explains. "Those are the kinds of things that we're trying to address here in the design of our system. We're not just building a sweetening room: ultimately, our plan is to evolve this into a practical, fully-digital audio post-production facility."

Digital Sound Effects

Clemens' noble aspirations notwithstanding, digital audio technology

Modern Videofilm's existing audio sweetening room also is scheduled for acoustic upgrade and refurbishment in the near future. Currently available equipment includes (*clockwise from top-left*): a 28-input Sony/MCI JH-500 Series console, UREI Time-Align monitor loudspeakers, and an Audio-Kinetics Q.Lock timecode synchronizer; Sony/MCI JH-110 eight- and four-tracks; a full outboard rack of signal processors; 3M/ITC NAB cartridge machines, Dolby noise-reduction units, UREI compressor-limiters, Eventide Harmonizer, Yamaha monitor power amplifiers, and a Sony/MCI JH-24 multitrack.

A central video prelay, layback and film-to-tape transfer center currently houses the facility's Sony PCM-3324 digital multitrack (*top-left*), plus Sony JH-110 eight-track and one-inch C-Format VTRs (*top-right*). A larger rack bay (*bottom-left*) houses additional VTRs and VCRs, plus video switching and distribution hardware. A companion video machine room on the facility's second floor (*lower-right*) links various production areas via audio, video and machine-control tie lines.

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seems to offer no immediate, cost-effective solution for dialog editing during electronic post-production. The few currently-available systems are far too expensive to permit the "horizontal spreading" of a project that Clemens cites as a major benefit of magnetic-film technology — even though they may promise flexibility and operational speed approaching that of the venerable Moviola. Clemens believes, however, that in the short term, he can win on another front: digital manipulation of sound effects.

"Both film and tape," he elaborates, "share a common problem: in either medium, the process of searching for sound effects is extraordinarily cumbersome and time-consuming. This is one place where digital audio technology can make a difference right now. With a 100 millisecond random access time — or even a second — you can just sit there and spot effects for a show, then have the computer assemble it and print out a cue sheet. We've got to get to that point in order to be competitive.

"So, we're looking at various devices for storing sound effects digitally, rather than having 1,500 carts on the wall. Right now, there are some

libraries available on Compact Disc, and a lot of people are recording sound effects in the field on F-1 format. So, the source material is starting to get there, and that means that we'll be able to build a fairly good digital effects library down the road."

Like many other audio post professionals, Clemens is considering currently available digital synthesizers as possible candidates for handling sound effects. "The Fairlight CMI [Series III], for instance, is a 16-channel device," he continues. "You could look at that as a multitrack, basically. But it also has the storage capacity to hold your library, giving you the ability to pull out an effect and put it exactly where you want it in relation to SMPTE timecode. And with the Fairlight, you can [manipulate] your effects with 0.01-frame resolution, or do pitch and time changes without running the signal through an outboard device.

"At the price, it becomes quite attractive: add a videocassette machine and a quarter-inch, and you're in business. With a skilled operator, you should gain a time advantage over a Moviola."

Jerry Clemens sees an immediate demand among producers for a useable random-access sound effects system. "What most of the people we've

talked to would like to be able to do," he relates, "is to spot a show and select their effects at the *same* time. With a random-access system, once you've done that, you've just about finished the show; you just run it and make any changes. Which is the type of system we're looking for. After all, that's not possible with a Moviola, and there's where you win: maybe, with the cost of the system, you can't spread it horizontally as much as you'd like to, but you make up for it. You could have three or four spotting stations, and do your assemblies overnight off the multitrack —be it analog, or digital-to-digital. There are some major possibilities for a system like that!"

The All-Digital
Post-Production Studio

Sound effects are not the only facet of audio post that Clemens plans to handle digitally. In keeping with Modern Videofilm's commitment to digital audio, Clemens and his staff vow that, within a year and a half, their audio post-production facility will be "almost totally digital." But a few hurdles still stand in their way and, when it opens, Modern's facility will be almost totally analog — with the sole exception of a CD effects library. *...continued overleaf—*

MODERN VIDEOFILM

Predictably. one substantial hurdle is the cost of establishing a fully digital facility: the current generation of digital multitracks are significantly more expensive than their analog counterparts. Inevitably. the investment is reflected in a facility's rates and, in the notoriously competitive world of audio post-production. the work tends to go to the house that offers the most attractive prices. Whether or not that facility can provide digital recording services is. for the most part. regarded as relatively unimportant.

Daunting as the cost factor may be. however. it is compounded by the current multitrack format war. Clemens explains: "I like the Sony [DASH-format] 24-track. but I sure wish we knew whether the format was going to stay the same. Is [Prodigital] 32-track going to become the industry standard? That's one of the things that's holding us up. You can't set up a very effective sweetening room with just one digital machine: you've got to have at least two machines. so that you can record on one and play back on the other — and that's a *substantial* commitment. If they change the format, you're up the creek!"

A special quality-control monitoring room enables MVF clients and operators to check the video and audio quality of film-to-tape transfers.

Despite such reservations. Jerry Clemens and Modern Videofilm are committed to an all-digital post-production facility. And while clearly foreseeing something of an uphill battle. they are nonetheless convinced that market forces will turn the tide in their favor. Clemens explains: "There are still too many people going around saying, 'It's *just* television; it's *just* a three-inch speaker.' But we know that's not true

anymore. especially when you consider the impact of multiple release formats. It used to be that, if you did a Music Special. it was just for broadcast. But now. you might see that show again in six months for reformatting for release on half-inch HiFi and Laserdisc.

"That's where we're going to be able to justify the cost of digital. Television isn't just broadcast anymore: it's also the Home Video market. so the product has to be as good as possible. After all. the person who buys a Laserdisc or half-inch HiFi VCR isn't just going to listen through his television: he's going to hook it up to his stereo! In reality. television specials now have to compete with the Compact Disc. and the audio quality has got to be *every bit as good.* Producers need to be aware of this."

By the same token. broadcast television also must compete with rental videocassettes of feature films. and Clemens believes that the audio quality of features may further influence the attitudes toward audio that prevail among TV producers. "Feature-film producers are going for higher quality sound every year," he offers. "so the film industry is going to help push television producers toward digital sound. When you can go and rent a cassette of *Raiders Of The Lost Ark,* where they spent a year and a half, or more, on the soundtrack. then television has got to get better. So. I think the film industry is actually going to *help* television go digital."

Clemens also counts on the recording industry. citing the increasing popularity of Music Videos and music-related specials. "The artists are going to want as much quality as they can get," he maintains, "because Stereo Television is *here.* They can demand it! And producers have to remember that a lot of those specials
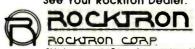
end up on the Home Video market.

"All of these factors are going to help the evolution of television sound. And, sure, it's going to cost the producer a little more but, down the road, he's not really going to have much choice in the matter. So, we're going to try to iron out the problems as soon as we can, in order to make digital sweetening a *viable* option."

Standardized Track Assignments

Whether audio is handled and recorded in digital or analog form, the central feature of electronic post-production is the use of multitrack transports — as opposed to Movieolas and a mag chain — to assemble and dub the audio tracks. And, since shows tend to move from one facility to another during post-production, the audio-post community has been gradually moving toward the adoption of standards that will bring a measure of consistency to the process. One such development is the standardization of track assignments for the 24-track format — a movement in which Jerry Clemens has been influential.
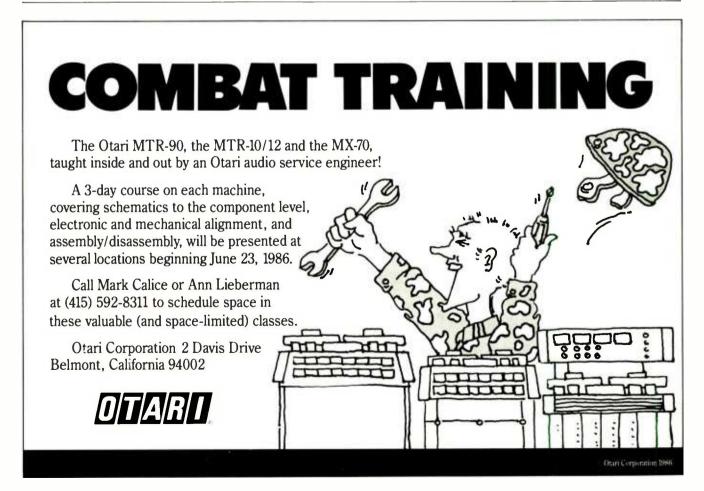
"We're pretty close to a standard," he relates. "At present, it appears that Glen Glenn, Compact, Complete Post and, I think, The Post Group will all be using basically the same track assignments, as far as what tracks are used for mix tracks, laydowns, SMPTE timecode, and so on. It's not a hard and fast format, but most houses are now using the same general assignments. And most houses are 24-track: we've pretty much abandoned 16, because it's just not enough tracks; you have to set aside eight tracks for mixing, two tracks for your videotape audio layback, plus SMPTE and 59.94. That doesn't leave much room to build anything on!

"As it has evolved, the standard assignments are as follows: track #1 through #8 are used as mixdown tracks; tracks #21 and #22 are the VTR [layback] tracks; track #23 is used for 59.94-Hz sync resolve; and track #24 is SMPTE timecode. The mixdown tracks are done in pairs: some people use track #1 and #2 for effects, and others #3 and #4, so that's not really standardized. In any case, the remaining tracks are used for building the various effects and dialog units."

Along with the independent post-production community, the NBC Network is reportedly close to adopting the same track-assignment standard, partially because of Clemens' influence. "When I was at Compact, which is across the street from NBC, I did quite a lot of consulting work for them. They've asked about formats, and we've tried to remain consistent. At present, NBC has some hardware problems that complicate the issue, but that will eventually be worked out."

"It's important to get some kind of a universally-accepted standard," Clemens concludes. "I suppose that if we all got on the phone, we could iron one out real fast, but communications between houses are fairly difficult: they don't want to talk!"

Communication problems notwithstanding, the audio post-production community seems destined to move increasingly toward electronic post techniques and, ultimately, toward digital audio technologies. While continued evolution of the hardware is both necessary and inevitable, Jerry Clemens and Modern Videofilm express the determination to take the plunge now — not only into fully electronic post-production, but also into the totally digital studio. "We really want to do it a little differently this time," Clemens says of the current planning at Modern, "and we intend to go more toward digital. As I say, it's expensive, but there are some overriding reasons to do it. And I think that the sooner we do it, the better off we'll be." ■■■

Photography by Rick Shaw and the Word of Faith Photographic Department.

FACILITY SPOTLIGHT:
WORD OF FAITH'S NEW AUDIO
PRODUCTION STUDIO

by Rick Shaw

It's unusual for a church to decide to build a network-quality production center, establish one of the largest closed-circuit satellite networks in the country, and then produce programs to air on it. Nevertheless, this is what Word of Faith Television in Dallas, TX, has done.

Robert Tilton, evangelist and president of the WOF network, determined that television could be more than simply an extension of his pulpit. Since religious programming is available in abundance, Tilton's organization began to think about using television to create unique and helpful programs, yet remain within the character of the network.

The flagship program, *Success 'N' Life*, was gaining major market acceptance. Most of Word of Faith's television programs are distributed by the use of its satellite up-link co-sited

with the studios. The actual "network" is comprised of over 1,800 affiliate churches, each of which is

— the Author —

Rich Shaw began studying music at the age of six, and his interest in keyboard instruments and electronics led to studying electronic music synthesis at Long Beach City college and California State University at Long Beach. In 1977, he accepted an audio position at PTL in Charlotte, NC, where he worked on live network television. Later, Shaw designed an audio production room for WPCB-TV in Pittsburg, and has served as free-lance engineer/composer/producer for numerous projects. In 1985, he co-founded Tri-Visions Communications, based in Atlanta, GA, with the role of executive director of audio productions. He is presently establishing Music and FX, a music production division of TVC, specializing in television themes, IDs, commercials, background cue music and electronic sound effects.

equipped with satellite dishes and wide-screen video monitors for their congregations. In this way, smaller churches would be able to view Word of Faith's live satellite broadcasts of important Christian artists and speakers. As well as the airing of *Success 'N' Life* via satellite, hardcopy tape distribution of the show is also provided through television and radio outlets.

In addition to providing what many would consider to be the best speakers available, Word of Faith also books top gospel recording artists. Many of these artists have become interested in creating their own videos at Word of Faith, but only if the proper equipment could be provided. For some time, multitrack recording trucks had been brought to record concerts performed in the 2,500-seat auditorium. While this provided a 24-track master

second floor → to engineering →

Hallway

RPG's

Speaker (4430R)

rack

RPG's

producers' desk

main video production console

50508

printer

trough

SSL

rack

triple rack

"VIP" viewing room

steps

Dubner

Ampex switcher

rack

4430L

360 Systems

ADO

ACE

Studer 900

JBL 4425

SSL computer rack

MTR-12c

MTR-90

desk

"curtain" wall

Audio Control

Sweetening Room

Video Production Control

Equipment Room

FLOOR PLAN AND EQUIPMENT LAYOUT OF NEW WOF FACILITY

WORD OF FAITH
— New Audio/Video Facility —

of the event, because of a lack of suitable equipment, little could be done with the tape afterwards. As is often the case, the concerts needed to be edited, sweetened and then packaged in a professional manner. The Home Video market also became an incentive to provide quality stereo mixes of the live concerts and similar presentations.

Design Concepts
Realizing that they were embarking upon a sizeable financial commitment, coupled with overwhelming local support for the idea of building advanced new facilities, Word of

Faith enlisted the services of Jerry O. Horstmann to provide initial designs for a new audiofor-video facility. Horstmann is probably best known as designer of the CBN Studios in Virginia Beach, VA, but has also been involved for many years in studio lighting and as the director of a number of network shows for CBS in New York. His experience in organizing and setting up large production and broadcast systems proved invaluable to Word of Faith. When the scope of the project began to be more fully understood, this writer was brought in as audio consultant by Horstmann.
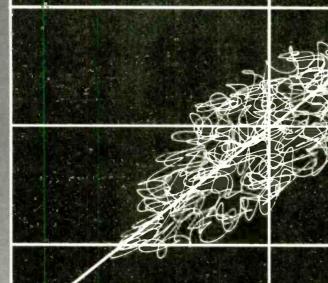
Initial meetings were held to determine what type of new programming was desired, and where the new control areas were to be constructed.

Originally, a new studio was to be built in a recently acquired warehouse facility in North Dallas, not far from the Word of Faith church building. This site proved to be too expensive to develop on the scale that the management wished: the budget would have climbed into the neighborhood of five to six million dollars. It was determined instead to provide a scaled-down version of the project, rather than using an inferior equipment package. A state-of-the-art audio and video package was proposed to management, and a detailed description of its advantages submitted.

The decision to go ahead with a quality equipment package made good sense, and proved the dedication of Word of Faith's management. Chief

Installing the Solid State Logic SL6000E mixing console.

Console shipping crate atop the new control-room slab.

The SSL console in its final room position, prior to the installation of monitor loudspeakers and central racks.

Video Production Control area prior to the fitting of various video switchers, monitor and routing controllers.

WORD OF FAITH
— New Audio/Video Facility —

engineer Dennis Ott, who had good systems experience gained at Turner Broadcasting in Atlanta, would oversee the technical aspects of the project.

Equipment Packages

The budget allowed approximately two million dollars to be spent on the audio and video equipment. Structural changes would also be needed to accommodate the new equipment within a second-floor area presently being used as control rooms, and located at the rear of a newly built auditorium used for church meetings and concert recording.

The selected video equipment included an Ampex production package: four VPR-3 one-inch video recorders; an Ampex Digital Optics (ADO); an ACE editing system; and a large video switcher with provisions for floppy-disk storage of set-ups and transitions. Also included would be a Dubner electronic animation/graphics computer, and a Utah Scientific integral routing-switcher system.

To ensure consistent quality from room to room, the entire audio/video complex would standardize on Asaca Shibasoku video monitors and JBL loudspeaker systems.

Audio signals would be split three ways via Jensen isolation transformers, prior to routing to three destinations: house mixer, monitor mixer, and TV mix position. It was determined that the house and TV mix positions needed the largest number of inputs. Signals were to be routed from the TV mix point to the house board so that VTR roll-ins could be heard by the audience during productions. At that time, house audio was being handled by a Soundcraft Series 400B console with 32 input channels. Since it was still providing adequate facilities, the 400B remains in that capacity. However, at least 40 inputs were needed for the TV mix position, because the Word of Faith's in-house band — which plays regularly for the Sunday church program — is also used as back-up for incoming artists.
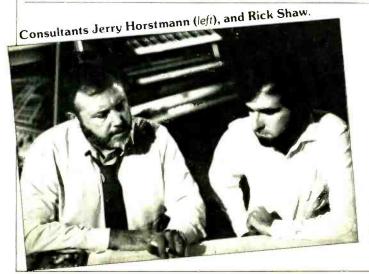
The monitor and TV mix positions were using Tascam Model 5 consoles with expander units. Although the Model 5s had performed admirably, they were straining under the load of new production demands. In light of the expanding programing needs, the TV mix positions were given first consideration.

A pair of audio control areas for video became an obvious choice, and one that proved to be the most cost-effective approach to handling daily audio tasks. The smaller room would be equipped with a 24-input Studer M-900 console to handle normal production, talk shows, voice-over work, and simple editing procedures. The M-900 was selected because it is well suited for live teleproduction, and is equipped with both AFL- and PFL-type soloing, quiet switching, and excellent channel equalization. The eight subgroups are usually routed into stereo and mono outputs.

Two JBL Model 4425 loudspeakers powered by a QSC Model 3200 amplifier are used in the Studer room for monitoring. An Otari MX-5050B handles recording and playback of material that needs to be inserted into programs. Also available are Studer A-710 and Nakamichi MR-1 cassettes.

Other equipment chosen for the smaller audio room included an MXR digital reverb, dbx limiter/compres-
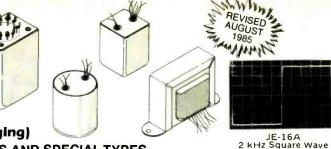

Consultants Jerry Horstmann (left), and Rick Shaw.

Mixing engineer Jay Yeary.

Audio Transformers
Choose from a wide variety of types and packages

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Wide bandwidth

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REVISED AUGUST 1985

JE-16A
2 kHz Square Wave

INPUT TRANSFORMERS AND SPECIAL TYPES

Model	Application	Impedance Ratio Pri-Sec	Turns Ratio Pri:Sec	20Hz Max Input Level[1]	Typical THD Below Saturation (%) 20 Hz/1 kHz	Frequency Response (dB ref. 1 kHz) 20 Hz/20 kHz	Band-Width[2] -3 dB @ (kHz)	20 kHz Phase Response (degrees)	Over-Shoot (%)	Noise Figure (dB)	Magnetic Shield[4] (dB)	Number of Faraday[4] Shields	Package[5]	PRICES 1-19	100-249	1000
MICROPHONE INPUT																
† JE-16-A / JE-16-B	Mic in for 990 opamp	150-600	1:2	+8	0.036/0.003	-0.08/-0.05	230	-8	<1	1.7	-30	1	A=1 / B=2	65.25 / 71.73	43.59 / 47.92	30.07 / 33.06
JE-13K7-A / JE-13K7-B	Mic in for 990 or I.C.	150-3750	1:5	+8	0.036/0.003	-0.09/-0.21	85	-19	<2	2.3	-30	1	A=1 / B=2	65.25 / 71.73	43.59 / 47.92	30.07 / 33.06
JE-115K-E	Mic in for I.C. opamp	150-15K	1:10	-6	0.170/0.010	-0.50/+0.10	115	-5	<7	1.5	-30	1	3	44.84	29.95	23.39
LINE INPUT																
JE-11P-9	Line in	15K-15K	1:1	+26	0.025/0.003	-0.03/-0.30	52	-28	<3		-30	1	1	105.75	70.65	48.74
JE-11P-1	Line in	15K-15K	1:1	+17	0.045/0.003	-0.03/-0.25	85	-23	<1		-30	1	1	42.69	28.53	22.27
JE-6110K-B / JE-6110K-BB	Line in bridging	36K-2200 (10K-600)	4:1	+24	0.005/0.002	-0.02/-0.09	125	-12	<1		-30	1	B=1 / BB=2	63.98 / 74.05	42.75 / 49.47	31.37 / 34.13
JE-10KB-C	Line in bridging	30K-1800 (10K-600)	4:1	+19	0.033/0.003	-0.11/-0.08	160	-9	<2		-30	1	3	43.45	29.03	20.03
JE-11SSP-8M	Line in / repeat coil	600/150-600/150	1:1 split	+22	0.035/0.003	-0.03/-0.00	120	-9	<3.5		-30	1	4	168.39	112.50	77.61
JE-11SSP-6M	Line in / repeat coil	600/150-600/150	1:1 split	+17	0.035/0.003	-0.25/-0.00	160	-5	<3		-30	1	5	85.11	56.86	39.23
SPECIAL TYPES																
JE-MB-C	2-way[3] mic split	150-150	1:1	+1	0.050/0.003	-0.16/-0.13	100	-12	<1		-30	2	3	36.22	24.21	18.89
JE-MB-D	3-way[3] mic split	150-150-150	1:1:1	+2	0.044/0.003	-0.14/-0.16	100	-12	<1		-30	3	3	63.35	42.32	33.04
JE-MB-E	4-way[3] mic split	150-150-150-150	1:1:1:1	+10	0.050/0.002	-0.10/-1.00	40	-18	<1		-30	4	1	98.99	66.13	45.62
JE-DB-E	Direct box for guitar	20K-150	12:1	+19	0.096/0.005	-0.20/-0.20	80	-18	<1		-30	2	6	45.46	30.38	23.71

1. (dBu) Max input level = 1% THD; dBu = dBv ref. 0.775 V
2. With recommended secondary termination
3. Specifications shown are for max. number of secondaries terminated in 1000 ohm (typical mic preamp)
4. Separate lead supplied for case and for each faraday shield
5. Except as noted, above transformers are cased in 80% nickel mu-metal cans with wire leads.

PACKAGE DIMENSIONS:

	W	L	H
1	1⁵⁄₁₆" Diam.	×	1⁹⁄₁₆"
2	1³⁄₁₆" × 1³⁄₁₆"	×	1⁵⁄₈"
3	1¹⁄₈" Diam.	×	1¹⁄₁₆"
4	1½" × 1¾"		2½" w/ solder terminals
5	1⁵⁄₈" Diam.	×	1¾"
6	1¹⁄₈" Diam.	×	1⁵⁄₁₆"

NICKEL CORE OUTPUT TRANSFORMERS[6]

Model	Construction	Nominal Impedance Ratio Pri-Sec	Turns Ratio Pri:Sec	20 Hz Max Output Level[7] (dBu)	across (n) windings	600Ω Load Loss (dB)	DC Resistance per Winding	Typical THD Below Saturation (%) 20 Hz/1 kHz	Frequency Response (dB ref. 1 kHz) 20 Hz/20 kHz	Band-Width -3 dB @ (kHz)	20 kHz Phase Response (degrees)	Over-Shoot (%)	Package	PRICES 1-19	100-249	1000
✱ JE-11-BMCF	Bifilar 80% nickel	600-600	1:1	+26	1	-1.1	40Ω	0.002/0.002	-0.02/-0.00	>10MHz	-0.0	<1[9]	7	65.36	43.66	30.12
✱ JE-11-DMCF	Bifilar 80% nickel	600-600	1:1	+21	1	-1.0	38Ω	0.004/0.002	-0.02/-0.00	>10MHz	-0.0	<1[9]	8	48.74	32.56	22.46
JE-123-BLCF	Quadfilar	600-600 / 150-600	1:1 / 1:2	+32	2	-1.1	20Ω	0.041/0.003	-0.02/-0.01	>450 / 170	-1.9 / -4.0	<1[8]	7	64.57	37.71	26.02
✱ JE-11SS-DLCF	Bifilar split/split	600-600 / 150-600	1:1 / 1:2	+27	2	-1.0	19Ω	0.065/0.003	-0.02/-0.01	>10MHz / 245	-0.0 / -2.5	<1[8]	8	46.38	30.98	21.37
✱ JE-11-ELCF	Bifilar	600-600	1:1	+23.5	1	-1.1	40Ω	0.088/0.003	-0.03/-0.00	>10MHz	-0.0	<1[9]	9	30.21	20.18	13.93
✱ JE-11-FLCF	Bifilar	600-600	1:1	+20.4	1	-1.6	58Ω	0.114/0.003	-0.03/-0.00	>10MHz	-0.0	<1[9]	10	23.66	15.81	10.91
JE-112-LCF	Quadfilar	600-600 / 150-600	1:1 / 1:2	+20.4	2	-1.6	29Ω	0.114/0.003	-0.03/-0.01	>450 / 205	-1.2 / -3.2	<1[8]	10	26.68	17.82	13.08
JE-123-ALCF	Quadfilar	66.7-600	1:3	+26.5	3	-1.3	8Ω	0.125/0.003	-0.04/+0.06	190	-4.6	<6[8]	8	44.09	29.45	20.32
JE-11S-LCF	Bifilar w/ split pri.	600-600 / 150-600	1:1 / 1:2	+30	1 (sec)	-1.7	63Ω	0.058/0.002	-0.02/+0.01 / -0.02/-0.05	>10MHz / 155	+1.1 / -4.1	<1[8]	8	44.09	29.45	20.32

6. Multifilar construction has no faraday shield: cannot be used as input transformer. All specifications are for 0Ω source, 600Ω load.
7. Max output level = 1% THD; dBu = dBv ref. 0.775 V
8. Source amplifier -3dB @ 100 kHz
9. Source amplifier -3dB @ 200 kHz
10. Output transformers are horizontal channel frame type with wire leads, vertical channel frames available. PC types available.

† IMPROVED PERFORMANCE   ✱ NEW MODELS

PACKAGE DIMENSIONS:

	W	L	H	Mounting Centers
7	1½" ×	2⁵⁄₁₆" ×	1¹⁵⁄₁₆"	2¹³⁄₁₆"
8	1⁵⁄₁₆" ×	1¹⁵⁄₁₆" ×	1⁵⁄₈"	2³⁄₈"
9	1¹⁄₈" ×	1¹¹⁄₁₆" ×	1³⁄₄"	2"
10	1¹⁄₁₆" ×	1⁷⁄₁₆" ×	1³⁄₁₆"	1¾"

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Prices shown are effective 8/1/85 and are subject to change without notice. Packing, shipping, and applicable sales taxes additional.

WORD OF FAITH
— New Audio/Video Facility —

sors, a routing switcher keypad, (part of the Utah Scientific system), and a multichannel RTS intercom station.

The other Audio Suite needed to handle live stereo musical events, as well as complex post-production sweetening and 24-track automated mixing. A console had to be chosen not only for its sonic quality, but also for its flexibility in broadcast and multitrack production applications.

This writer also felt that the selected console should have individual dynamics sections on the inputs to help handle the unexpected during live production. Available gating on the inputs would also help eliminate noise in the auditorium, as well as leakage during live recordings of musical groups. Another benefit of multichannel gating is that it provides automatic noise reduction when working with multitrack and/or a large number of keyboards. We looked at automated consoles suitable for live broadcast and post work, and ones that incorporate built-in dynamic sections.

The final selection was a Solid State Logic SL6000E console. Originally, to help save on expense, the console was

Close-up detail of the completed Video Production Control Room, showing the various Preview and Program video monitors, central switcher and routing panel, plus controls for a variety of graphics and visual special-effects units.

configured with a 40-input mainframe loaded with 32 modules. But, even as the console was being constructed, elements were being added to the WOF productions, with the result that the console would be too small by delivery time; eight additional modules were fitted prior to delivery. Also included in the order were several options: Total Recall automation; a Master Transport Selector, Events Controller, Synchronizer Controller, and Screen Printer.

Additional equipment for the second suite included an Otari MTR-90 24-track; an Otari MTR-12c (the "c" denotes a center-stripe timecode track); JBL 4430 monitors driven by a UREI 6500 power amp; Klark-Teknik stereo third-octave equalizers; an Adams Smith Series 2600 timecode synchronizer; Lexicon Model 200 digital reverberation; an Aphex Compellor; Eventide Harmonizer; two Nakamichi MR-1 cassette recorders; and a 360 Systems digital keyboard for creating in-house music beds. (At the time of writing, a Fairlight CMI Series III digital synthesizer is being considered.)

Construction and Room Layout

The second floor area used by Word of Faith for its technical area was not originally designed for this purpose. Structurally, there was some concern to whether the area could safely support the new loads. While the new one-inch VTRs presented a weight problem, the SSL console had no other real competition when it came down to sheer mass: it measured out at 10.6 by four feet, and weighed about 1,800

....continued overleaf --

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SONY
Professional Audio

WORD OF FAITH
— New Audio/Video Facility —

pounds. No area of the second floor level had been designed to handle the combined weight of the new audio and video equipment.

In addition, acoustical treatment of the second floor had never been taken seriously. Its location in the same auditorium structure had always created severe isolation problems, and sound tended to leak from the auditorium into the control areas. People talking in the control rooms could often be heard in the auditorium during quieter moments. The structural and sound-isolation problems had to be taken care of before any of the installation could begin. All of these conditions were taken to Jerry Horstmann for approval of acoustical design work.

During this time, Word of Faith's engineer, Jimmy Kirkpatrick (now working for Midcom in Dallas), knew of the Joiner-Rose Group in Dallas, a company that specializes in room acoustics. The two of us had a meeting with Russ Berger, senior consultant at Joiner-Rose. We were excited about Berger's approach to acoustics and the possible use of Reflection Phase Grating (RPG) technology to improve

sound diffusion in the audio control rooms. (Berger also served as audio acoustics designer at TeleImage and Dallas Sound Labs; TeleImage installation also incorporated an SSL in its audio post room.)

Since so many problems existed with the layout and load-bearing capacity of the second-floor area — which was the only space available to us for the project — it had to be completely torn down and rebuilt. This decision held up the project considerably, and was one of the major reasons that this writer became involved with the project for over a year and a half. The floor was taken down in halves so that all productions could continue during the reconstruction. Each section was rebuilt using new and heavier steel beams and supports. As recommended by Berger, a new four-inch concrete slab was poured, an addition that radically improved the feel of the whole area.

After the first half of the second floor had been redone, all of the equipment was moved over as the other section was demolished. Since none of the passageways leading up to the second floor were large enough to provide sufficient access for the SSL console, we took the opportunity during the construction to fork-lift the

desk up through the removed half of the floor. Although the load-in went without a hitch, watching the 1,800-pound, $240,000 console suspended in mid-air during its trip to its new home was not without tension!.

The audio control areas were specified to have slab perforations around each room for sound isolation. Each room slab was poured separately, and a one-inch gap filled with fiberglass surrounds each slab. To improve isolation between the auditorium and the technical areas, a "curtain" wall was constructed. Actually, this wall is two separate walls not bridged with any rigid connection, a dense, mineral sound-attenuation blanket being added to fill the space between the two.

Acoustic Considerations

For the amount of equipment they are designed to house, each audio room is rather small. We lost some space to an adjacent equipment room that houses the SSL computer rack and screen printer. To overcome the space problem, Berger used RPG diffusers to provide good diffusion of audio energy, and to make the room sound bigger than it really is. Both audio rooms were designed in an LEDE-style (Live-End/Dead-End), with the front half of the room being as anechoic as possible, and the rear utilizing the RPG diffusers. The front half of the room was made acoustically dead by using a stretch fabric wall treatment applied by Professional Wall Upholstering, Inc., Dallas.

To improve the perceived "tightness" of sound in the room, the JBL cabinets were supported on cement-block pillars hidden within the monitor wall. Between the speakers, two racks house the color video monitors: one for the SSL computer displays and Total Recall; and the other for video playback via the routing switcher keypad.

A V-shaped viewing room located at the back of the main video production/editing control area was designed to double as a voice-over booth, and became known as the VIP Viewing Room. The booth was also treated on its own separate slab using the same methodology as the audio control areas. Ceilings in all of the audio areas were suspended by special mounts hung from building steel, and are hard ceilings that have been gasketed around the edges with no lay-in tiles. In the places where sound absorption was required, a matching stretch fabric treatment was applied. (Stretch fabric wall treatment also helped with the acoustics in the VIP Viewing Room.)

Because it was considered impor-

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WORD OF FAITH
— New Audio/Video Facility —

tant that the ceiling not be penetrated while recessed lighting fixtures, track lights were used in all of the sound-proofed areas. A double-circuit track system allows various areas to be lit as desired.

Cabinetry was then designed by this writer for all audio and video technical areas. This task proved to be particularly difficult for the video areas, because of the lack of standard equipment sizes. Video control panels can come in a variety of shapes and sizes; sometimes there is little thought by the manufacturer about how the user will mount the unit in an ergonomic fashion. Bob Oliver, who has his own cabinetry manufacturing

facilities in Chattanooga, TN, built all of the custom consoles for Word of Faith. We experimented with Color-Care Formica, a material that displays no dark edges after it is routed, since the color goes all the way through the plastic.

Equipment Interconnects

Oak wiring troughs were constructed in both audio control areas to service tape machines and equipment racks, a design concept that helped to conceal all audio cabling. The trough located beneath the SSL also protected the many rather vulnerable ribbon cables running to the console from the computer rack.

Interfacing the various tape machines to the synchronization system was carried out by SSL. The Otari

MTR-90, MTR-12c and a Sony BVU-800 U-Matic VCR are normally used during any sweetening assignment. The SSL Master Transport Selector allows an engineer to designate which machine will become the master, or which machines will be called upon to slave to the master. A set of remote controls located just below the computer keyboard on the SL6000E can be used for any of the machines. The SL6000E allows three stereo mixes to occur at the same time: Mix "A" could have just stereo orchestra, Mix "B" English dialog and the stereo orchestra; and Mix "C" French dialog and the stereo orchestra. All of these stereo signals can be routed independently to individual machines for recording, and a 24-track master made at the same time by routing off

MUSIC RECORDING AND POST-PRODUCTION AT WORD OF FAITH TELEVISION

During the past two years, several notable video presentations have been shot in the Word of Faith auditorium. Artists such as Debby Boone, The Imperials, Phil Keaggy, Russ Taff, White Heart, and others have appeared live in concert via the satellite network. *The Imperials' 20th Anniversary Concert,* directed by Michael Stephens, is now available for the Home Video market, and was recorded prior to installation of the SL6000E console. Paul Christensen from Omega Audio brought in his 24-track remote truck, equipped with two Otari MTR-90s and an API console, for the TV/recording mix.

Word of Faith's new video production facilities allow in-depth post-production of concert material. Through the use of the Dubner graphics generator, ADO, and other effects, WOF has achieved a commercial "look" that many consider to be unique in the category of "religious" presentations.

The Word of Faith band, directed by Art Osborne, appears regularly on much of the programming. Art Osborne, formerly the musical director for *Captain Kangaroo,* also plays sax, flute, and arranges most of the music performed by the band made up of professional studio musicians. Since the installation of the Solid State Logic console, Osborne plans to use the band to create original music beds for in-house video presentations. Osborne composed the theme to the syndicated new version of *Success 'N' Life,* hosted by Robert Tilton.

There is also an interest in music production through digital synthesizers and samplers. Two music packages were produced by this writer for Word of Faith's satellite network, the latter of the two packages being created on the new recording equipment with the use of the following musical instruments: a 360 Systems digital keyboard with MIDI interface (loaded with 18 instrument sounds); Roland JX-8P; Ensoniq Mirage; and a Sequential Drumtracs digital drum computer. A Yamaha DX-7 is also used regularly for various productions.

There is a current interest at WOF in becoming more involved with setting up a sophisticated MIDI/sampling keyboard system to be used for music production, sweetening and scoring tasks. The proposed setup will use a SMPTE-based MIDI sequencer that will provide addressing information for the SSL computer, so that the automated mix functions could be used in a "tapeless" system. Mixing engineer Jay Yeary will be setting up his portable PCM recording system to capture sound effects in the field for later use on various sweetening assignments. □□□

— Russ Taff —

— White Heart —

— Phil Keaggy —

The author seated at the SSL SL6000E Production Console.

The routing switcher enables the operator to assign the 16 SSL inputs to receive audio from any source in the house, up to 16 at a time, or as eight stereo pairs, thereby creating instant access to VTR audio playback, or any other source without patching. A break-away feature allows the engineer to view one video source while tracking another. It was decided that the Utah Scientific routing switcher would be set up to operate in either stereo or mono mode. Each technical area has its own dedicated routing switcher keypad.

During the time that the rooms were being built, two new audio engineers were employed by Work of Faith to help handle the work load. Chris Godwin assumed the responsibilities as Audio Supervisor, and Jay Yeary, with nine years of live-television audio experience, assuming the position of Mixer One. Together, they continue to make the system fly.

By specifying high-technology production equipment for Stereo Broadcast, videotape distribution, and radio syndication, Word of Faith Television represents an excellent example of the forward-thinking movement taking place in the field of audio production for broadcast. ∎∎∎

the appropriate buss assignment switches provided on each channel module.

Word of Faith's console was configured so that inputs #1 through #24 appeared on the left side of the main facilities panels, and inputs #25 through #40 on the right. Line inputs on the left side were assigned to the MTR-90's replay outputs, and the remaining 16 line inputs on the right side to outputs from the routing switcher — a console configuration that allows good habits to be formed.

Photography by Elizabeth Annas

"The Computer Age" has certainly brought joy to many in the recording industry. Computerized consoles, synthesizers, auto-aligning tape machines, and digital signal processors have all contributed to the ease of operation of our daily recording routines. Or have they? The advent of such technology has brought with it the difficult problem of interfacing different pieces of computer-controlled hardware. More often than not, we are dealing with equipment from different manufacturers, each of whom may have their own way of addressing the relevant communication parameters. Whether one is using MIDI, RS-232/422 serial ports, or some type of sync-pulse, in order to ensure an operational system, we must be careful to maintain the proper interface parameters.

This article deals with the experience of interfacing a small arsenal of audio, video, processing and performing equipment for the purpose of recording a film soundtrack. At first, it appeared to be the perfect opportunity to test several new pieces of interrelated gear for an *R-e/p* Operational Assessment. Instead, it became an exercise in patience and cunning to complete the film on schedule.

The venue we selected for this in-use assessment was one that is becoming increasingly more familiar, namely a personal-use/home studio. An increasing amount of the work for industrial films, shorts, radio and TV commercials has shifted out of the larger, more expensive, studios and into the potentially more cost-effective home studio environment. This migration is due in part to the fact that the recording equipment available to lower-priced production facilities is becoming more and more sophisticated. The audio equipment used in this review was chosen specifically to emulate a limited-budget recording situation, where music had to be synchronized to picture. It should be noted that the line between "pro" and "semi-pro" gear — whatever those designations have come to mean in today's diverse recording world — is becoming less distinct. Several manufacturers are currently offering tape machines, mixing consoles, effects processors, and other items of recording hardware with specifications that allow the use of either -10 dBV or +4 dBm levels, thus making system interfacing and upgrading both easier and more desirable.

The equipment under evaluation included the Synchronous Technologies SMPL Chase Lock System; the Lexicon PCM-70 digital effects pro-

COLLECTIVE EQUIPMENT ASSESSMENTS IN AN ELECTRONIC-MUSIC PRODUCTION FACILITY

- **Synchronous Technology SMPL Chase Lock MIDI and Timecode Synchronization System**
- **TASCAM Model 388 Studio 8 Combination Mixer/Eight-Track Recorder**
- **Lexicon PCM-70 Digital Effects Processor with Dynamic MIDI™ Control**
- **Southworth Music Systems Total Music MIDI Sequencer for the Apple Macintosh**
- **Opcode Systems MidiMac Sequencer Software for the Apple Macintosh**

Reviewed by Bob Hodas and Denis Hannigan

cessor; the Tascam Model 388 Studio 8 combination eight-track console and tape machine; MidiMac Sequencer software from Opcode Systems; and Bill Southworth's Total Music MIDI software. Additional equipment included a Yamaha DX-7 FM synthes-

— the Authors —
Denis Hannigan is a musician/composer with extensive experience in synthesis and horns. He has scored music for album, broadcast, and industrial projects, and is currently working on a feature film entitled *Thrashin'*.
Bob Hodas is an independent engineer with backgrounds both in the studio and on the road. Included in his credits are sessions with such groups as The Village People, Mickey Hart, The Doobie Brothers and Marty Balin. He also acts as a consultant to several U.S. and Japanese audio manufacturers.

izer, plus two TX-7s; Oberheim Expander; Roland JX-8P; LinnDrum; NEC N895EU half-inch VHS HiFi videocassette recorder; Apple Macintosh computer; and two video monitor/receivers. Interface boxes included the Roland MPU-104 Input Selector, MPU-105 Output Selector, Korg KMS-30, Roland SBX-10, and J.L. Cooper Micro Sync. As will be appreciated from the accompanying photographs, we assembled enough equipment to fill a typical spare room or den, and created quite a snake pit with the cables.

The studio was located in the home of composer/musician Denis Hannigan, in Beverly Hills, CA; Denis owns the musical instruments and interfaces listed above and, in order to work more efficiently with sequencing and

COLLECTIVE IN-USE ASSESSMENTS

composition, recently added an Apple Macintosh personal computer. So that he would be completely familiar with all aspects of the software program, Denis had secured a copy of Southworth Music Systems Total Music about a week prior to the start of the scoring project. Denis and I have done a number of film projects together. During that time, we have worked out a systematic approach that keeps our sessions running smoothly and efficiently. This familiarity would prove to be invaluable as this evaluation progressed.

Synthesizer and Computer Routing

When I arrived, the first thing we did was to define the routing of the computer and synthesizers via the proper interfaces. Denis wanted the system setup to perform the following functions, without having to repatch any MIDI cables.

1) He needed to operate the keyboards, even when the computer was not running any sequencer software;

2) Assign MIDI to the individual keyboards, so that he could listen to all or any individual keyboard as desired;

3) Assign any individual keyboard MIDI Out to the computer for making voice-data dumps to floppy disk; and

4) Avoid MIDI time delay by not utilizing the keyboards' MIDI-Thru jacks.

All of the above requirements were satisfied by using the Roland MPU-105 Output Selector, MPU-104 Input Selector, and a simple switch that I built from about $10 worth of parts. The MPU-104 was used to route MIDI
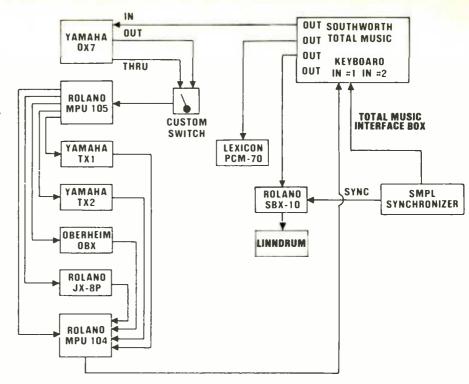
FIGURE 1: BLOCK DIAGRAM OF SYNTHESIZER'S ROUTING AND SEQUENCER MIDI INTERCONNECTION

Outs from the keyboards to the Mac, and the MPU-105 to connect the DX-7 controller to the keyboards' MIDI Ins. (If you think this sounds backwards and confusing, don't worry, I think so too! The MPU-105 is really designed to route the MIDI signal to individual inputs so why, I wonder, is it called an "Output Selector?" The same logic applies to the MPU-104.)

The home-made switch connected either the DX-7's MIDI Out or Thru to the MPU-105's MIDI In, a setup that allowed Denis to switch between MIDI Out (DX-7 control) and MIDI Thru (Macintosh/Total Music control). Figure 1 shows a block diagram of the final routing at the completion of this project, and should give the reader a clear picture of our connect logic.

Since the LinnDrum is not MIDI controllable, it was connected to the Mac via a Korg KMS-30 (which converts a MIDI signal into 24-pulse sync on a DIN connector), and then to a J.L. Cooper Micro Sync (which converts the 24-pulse DIN to 48-pulse sync via phone jack needed by the LinnDrum). It should be noted that we realized that the KMS-30 was also capable of producing 48-pulse sync, but we needed a phone-jack input to the LinnDrum, and not a DIN connector. All MIDI connections to and from the Mac were routed through the Total Music Interface box, which appears to be well constructed and versatile with its four MIDI Outs and two Ins.

The configuration described above should not be taken as the "ideal" interface method, since there are many selector boxes and converters on the market, and new ones popping up every day as the mysteries of MIDI interfacing are unraveled. Detailed here is simply the equipment we had to work with at the time and, as you will see, our setup changed as we found and solved new problems.

Audio and Video Connection

The next step was to hook up the

DETAILED PRODUCT DESCRIPTIONS

• SMPL Basic Chase Lock MIDI and Timecode Synchronization System; SMPL Lock (with rack unit containing control outputs and updated software V1.0).
Pro-User Price: $750; $1,495, respectively.
Manufacturer: Synchronous Technologies, 1020 West Wilshire Blvd., Oklahoma City, OK 73113. (405) 842-0680.

• TASCAM Model 388 Studio 8 combination mixer/eight-track recorder
Pro-User Price: $3,995
Manufacturer: Tascam, 7733 Telegraph, Montebello, CA 90640. (213) 726-0303.

• Southworth Music Systems Total Music MIDI Sequencer for the Apple Macintosh.
Pro-User Price: $489.
Manufacturer: Southworth Music Systems, Inc., 91 Ann Lee Road, Harvard, MA 01451. (617) 497-7522.

• Lexicon PCM-70 Digital Effects Processor with Dynamic MIDI™ control.
Pro-User Price: $2,295.
Manufacturer: Lexicon, Inc., 60 Turner St., Waltham, MA 02154. (617) 892-6790.

• Opcode Systems MIDIMAC Sequencer software (V1.0) for the Apple Macintosh (soon to be released V2.0 with enhanced step-entry editing).
Pro-User Price: V1.0 $150; V2.0 $200.
Manufacturer: Opcode Systems, 1040 Ramona, Palo Alto, CA 94301. (415) 321-8977.

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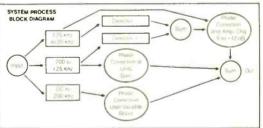
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SYSTEM PROCESS
BLOCK DIAGRAM

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audio and video gear, and get to work. Synthesizer outputs were connected directly to the Studio 8's channel line inputs, along with the PCM-70 stereo returns. Echo send output from the Studio 8 was fed to the PCM-70's single input at line level. The PCM-70's MIDI In was connected to one of the four MIDI Outs on the Total Music interface box. I now approached the SMPL Chase Lock unit, only to find that no description had been included in the manual of front-panel access ports and connectors. Well, having some experience with audio-video interfacing, I jumped in and connected the Chase Lock to the audio and video transports. The panel itself turned out to be fairly straightforward: the Studio 8's back-panel accessory access jack was connected to the Chase Lock Slave port; the Slave Timecode Input to the Studio 8's channel #8 direct out; and the Master Timecode Input to the VCR's right-channel audio output.

The Chase Lock has two timecode outputs for simultaneously recording code onto two machines, which were connected to the VCR's right-channel audio input, and the Studio 8's channel #8 input. The Studio 8 is equipped with a switch that defeats the dbx noise-reduction circuit for track #8, while working with timecode or sync-pulse tracks. (Noise reducing such tracks is unnecessary, and can lead to problems during lockup.) Since the

NEC half-inch VCR is a consumer unit, it has no remote transport control capabilities, and so nothing was connected to the SMPL System's Master Control port. With all of the audio and video hardware hooked up, I made a quick call to Synchronous Technologies to make sure that I wasn't about to blow anything when power was applied! Having assured myself that the input/output configuraion was correct, we were ready to roll tape.

Timecode Synchronization

A videocassette copy of the film workprint arrived at the studio with a burned in timecode window, but no code recorded on the audio track. Denis spotted the tape copy for his music cues, and calculated tempos and durations according to this visual readout. I proceeded to stripe timecode from the SMPL System onto the relevant audio and videotapes for sync lock. To allow us to properly calculate the punch-in and -out times for matching cues. I then ran the video-cassette and calculated the timecode offset between the striped code and the visual SMPTE timecode display.

Since the SMPL System timecode had been laid down after the visual code, at the very extreme we would be half a frame off from the latter. (Of course, if the the director had supplied us with a matching audio timecode track, we could have had exact sync!) Even though the SMPL System does not regenerate timecode, if you start with a good audio code track, it can

usually be copied a generation down without significant problems. (I am not recommending this practice; sometimes it is the only viable alternative.)
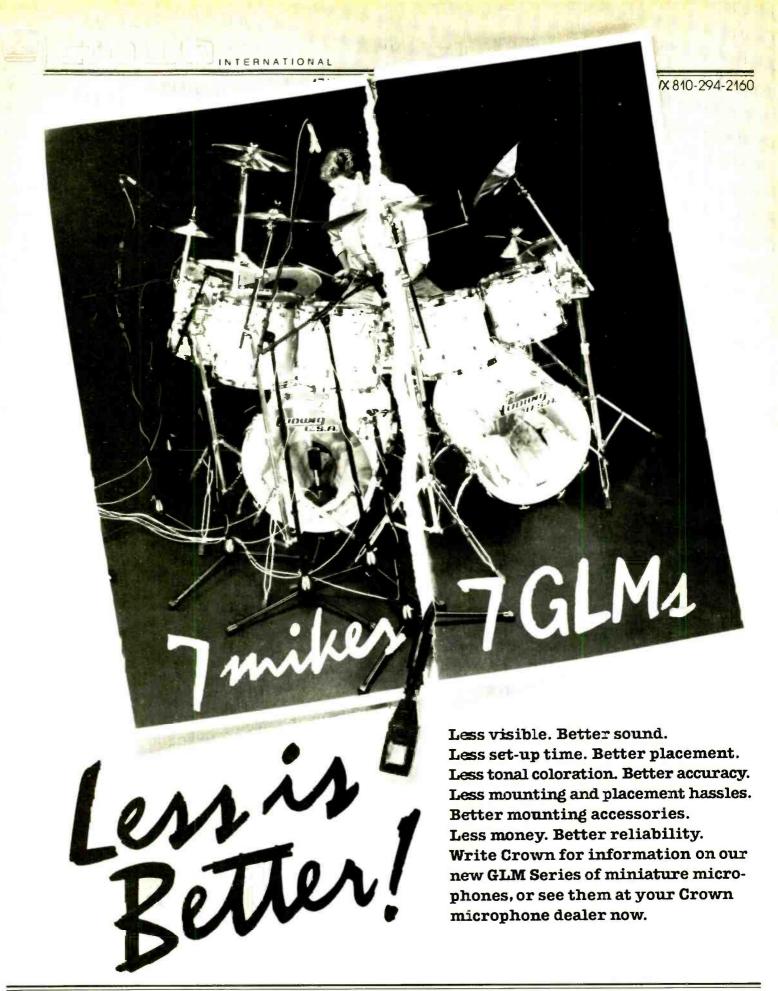
Having striped both tapes, I proceeded to follow the manual's description for calibrating the Chase Lock. This proved to be a problem, since the manual talks about adjusting calibration only in the Lock Mode, and I planned to operate the machines in the Chase Solo Mode. I thus had some difficulty with certain controls, and didn't achieve the desired results in locking together the two machines. After a good deal of frustration and experimentation, another call to the factory was in order, at which time they explained that Sychronous had omitted mentioning the Chase Solo calibration procedure. I was able to calibrate in the proper mode, and everything seemed to work fine. The calibration procedure was simple and easy to follow, and my only recommendation would be to see better quality pots installed in the unit.

The actual synchronizing of the Studio 8 and NEC half-inch VCR worked quite well — both machines locked within four seconds of passing the SMPL pre-roll cue point — although the procedure for setting this up was a bit time consuming. The master VCR had to be run and then put into pause at the appropriate cue point. The timecode location was then assigned to the Cue Window and, by hitting the SMPL transport key twice, the slave would locate to within five seconds prior to the cue point. Punch-

The Synchronous Technology SMPL Chase Lock system (seen with reviewer Bob Hodas *above*) comprises a modified Commodore VIC-20 PC running custom software (*right*) connected via an interface card to an external transport, timecode, sync and MIDI controller (*below*).

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in and -out timecode locations were then assigned. When the master machine was started, the slave would also start and lock, and the SMPL System would then perform its designated record in/out tasks.

We quickly discovered that things really moved along much faster if the VCR was temporarily omitted from the system while recording the audio tracks. If we had pre-calculated the cue points, and the SMPL System handled all that it was supposed to, after the audio had been recorded the two machines should sync perfectly, with the audio in its appropriate place. In fact, once we solved a couple of problems we were having with the synthesizers, such a scheme became our recording procedure.

Synthesizer Time-Lag Problems

The problem, to be exact, was that the synthesizers were lagging behind the LinnDrum by about 20 to 30 milliseconds. Since something was causing the synths to run more slowly, we started eliminating factors as we slowly broke down the system. After a

while, I began to suspect that the MPU-105 was slowing down the speed with which the Total Music program was communicating with the synthesizers. However, when the Mac was replaced with a Roland MSQ-700 sequencer the problem disappeared, which indicated that the problem might have been caused by Total Music itself. Denis called Southworth, and was given a number of different approaches to try. Although most of these fixes were unacceptable, one of them did seem to help considerably. Denis also made some calls around town, and the general consensus was that the problem actually stemmed from the interface of the KMS-30 and Micro Sync.

When these two devices were replaced by a Roland SBX-10, our problem was solved. It appears that the SBX-10 reads MIDI data at the same rate as the other interfaces, and the LinnDrum ended up in perfect sync. (An added bonus is the fact that the SBX-10 is cheaper that the KM-40 and Micro Sync combination.)

Once we got down to recording, we hoped that things would go relatively smoothly. At first, eight cue points were recorded onto the SMPL System's Events Page, which could then

be easily transferred over to the Rehearse Page as we went along. After each cue was transferred, the record-in and -out times were programmed into the unit, the Studio 8 would roll back to five seconds before the cue point, and we were ready to go. The method at this point was for me to to start the Studio 8 rolling, and for Denis to then enable replay of the Total Music sequence. When the SMPL System detected the appropriate cue point, it would send a MIDI signal to Total Music to begin outputting MIDI data to the bank of synthesizers.

As smoothly as our technique worked, however, we had struggled to reach this point in the proceedings for the following reasons.
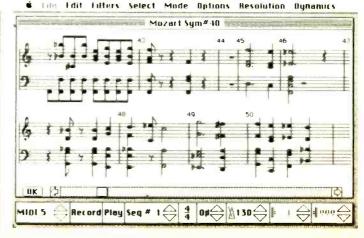
• Every time we started it, the SMPL System Chase Lock sent a pulse through the system that would trigger Total Music. For this reason, we always started Total Music *after* the tape was rolling, and the software sequencer would then wait for the cue pulse. With some of the other problems we were having, it took a while to figure out this one. The Southworth Music interface box for the Macintosh is equipped with LEDs that show the presence of MIDI signals, which

Reviewer Denis Hannigan (*below left*) at the Apple Macintosh PC running Southworth Music System Total Music MIDI sequencer/recorder software, which includes a custom interface unit for synthesizer control (seen *below right* beneath the Mac and external disk drive). At the foot of the page are reproduced screen dumps of a MIDI data stream (*left*) that enables close examination and editing of individual MIDI events, and a Grand Staff screen (*right*) that provides details of recorded MIDI data in standard music notation.

proved to be invaluable for tracing down problems during the session. (The pulse problem, I am told, is currently being corrected by Synchronous Technologies.);

• Total Music's MIDI Song Pointer did not work with the SMPL System. (Song Pointer is part of the MIDI spec which, based on information received from the timecode track, will locate sequencers to the proper bar and note — an absolutely essential feature for fast autolocates and overdubs in the middle of a song.) We spent hours trying different combinations of options, but to no avail. Denis finally called Southworth Music, only to hear that "Yes, this is a problem, and we are working on the solution." (The Total Music manual does state that the program works with the Roland SBX-80 Sync Box, although we did not have the opportunity to verify this statement.)

• I had forgotten just how slow punches are on tape machines that run at 7.5 ips! I must credit Tascam with mounting the Studio 8's heads very close together, but the slow tape speed still hampers a tight punch; I'm just spoiled by 30 ips. This certainly did not prove to be a major problem, however, since the solution was to program the SMPL System to punch-in several frames early. The tech-

nique worked marvelously, and I was impressed with just how tight the punch-ins could be — the unit allowed us to achieve consistent, repeatable, clean punch-ins. Punch-outs on the Studio 8 are much slower, however; although we could get better resolution with the SMPL System, we could not get out of really tight spaces.

With all these problems resolved, we went on to establish a recording groove and zipped through the film's soundtrack. Since Total Music's Song Pointer didn't work with our system configuration, we treated each cue as a separate song, and simply triggered the sequence from the beginning. No overdub punches were performed.

When all the recording had been completed, I calculated the offset necessary between the video-window timecode and that recorded on the tape transports. Since the SMPL System prefers to see a one-minute timecode leader on the slave machine (to prevent overshoot during fast searches), and the videotape did not have a one-minute blank prior to the start of the film, I had started recording timecode on the NEC VCR about 15 seconds *after* the Studio 8 slave machine had started. This offset was programmed into the SMPL System's offset window, and we rolled the NEC master. Well folks, every cue was

right where it belonged in the video. The project was complete and Denis and I broke out the soda pop to celebrate.

System Components

Let's look individually at the different items of equipment that were used to record the film soundtrack. One piece of gear that gave us absolutely no problems was the Lexicon PCM-70, a mono-in/stereo-out digital reverb and effects processor with Dynamic MIDI control. To save time in the mix, all tape tracks were recorded with effects; the PCM-70's versatility was up to the job. First let me say that this unit is capable of producing a very good sound. The software programs are warm and rich, and the delay sounds like a sample of the original signal. There are a total of 36 preset programs to get you started, and room to store an additional 50 user-defined registers. Seven dedicated MIDI preset programs are also available. The three banks of preset effects are Chorus and Echo (10 settings), Multiband Delays (nine), and Resonant Chords (six), while the three banks of preset reverb programs comprise Concert Halls (three), Rich Chambers (five), and Rich Plates (three).

Front-panel controls on the PCM-70 are simple and easy to learn, and all

COLLECTIVE IN-USE ASSESSMENTS

visual information is provided on a legible 16-character, alphanumeric LED display. Programs and parameters are stored in rows and columns, emulating a spreadsheet set up. What I find so amazing about this unit is the sheer power in the number of adjustable parameters provided for the operator. The PCM-70 is not a unit on which you simply dial up a great setting by accident; you will have to know a bit about reverb theory and parameter manipulation as you scroll through the variables. However, Lexicon has provided the user with full support regarding parameter description and usage in the user manual.

Denis was really happy with the degree of program and dynamics control available through MIDI, as well as being impressed with the sound quality. He manipulated patches and parameters in real-time with a Yamaha DX-7 mod wheel and after-touch control. Having stored these MIDI changes on Total Music, the software "played back" the PCM-70 as if it was just another sequenced synthesizer. The potential for storing effects in this manner is enormous, and can save a lot of time both on the road and in the studio. Lexicon has a very strong product in the PCM-70.

The Lexicon PCM-70 features 36 factory presets and 50 user registers, plus Dynamic MIDI for real time control via an external mod wheel and/or after-touch.

Tascam's Model 388 recording combo also proved to be completely trouble-free during the course of our session; considering its low price and targetted market, it was a good performer. The Studio 8 is aimed at the broadcast production and demo markets, and seems able to provide the necessary functions to turn out product. A combination eight-input console and eight-track on quarter-inch tape transport utilizing defeatable dbx noise reduction, the deck features a two-head design, and thus is always ready for sync recording. The system is set up to interface with both +4 dBm and –10 dBv input/output levels, and the back-panel arrangement is clean and logical. Almost any access to channels or busses that you would want is provided; there will be no interface problems with this unit.

A big advantage over earlier Tascam recorders is the provision of an insert switch that allows auto-switching between tape and input signals during punches. Transport controls are also convenient, and feature Return-To-Zero, Search-To-Cue, and an Auto-Repeat function. Varispeed is available (±15%) and, of course, external control of capstan speed and transport functions. What particularly amazed me about the recorder's electronics was that I could really slam the tape with level, without it over-loading. Normally, with a narrow-gauge multitrack, I would roll off a lot of bass before recording, but found this unnecessary when working with the Studio 8.

On the console side of things, each input channel has both mike- and line-level inputs with mike trim; three-band EQ with center-frequency selection; plus effects (pre-) and auxiliary sends (pre- or post-fader). Two pannable effect returns and send masters are available. The pannable monitor section works fine and, with a little creativity, can be used for additional sends.

I didn't find any major problems with the mixer or multitrack, and the extra features I'd like to see would probably add considerably to the price. For example, I would like to see at least two more channel inputs for those of us building up our arsenal of synths and rhythm machines. A channel line trim would also be nice, as would separate volume pots for the monitors and headphones. Oh yes, and a high-pass filter on the channel modules. Such requests come from my major studio multitrack work and, I would be the first to acknowledge, are a bit unreasonable considering the Studio 8's intended market. All in all, we got a good sounding film track with the system and that, after all, is

the bottom line.

Oh yes, I must mention the superb manual that comes with the Studio 8. It would take someone that has never seen a tape recorder before, all the way through the basics of recording operations, right into learning how to do your own maintenance. In fact, the manual is a perfect combination of simplicity and techology; I take my hat off to Tascam for a spendid achievement.

The SMPL Chase Lock is the lowest-priced synchronizer on the market, and performed all of its functions correctly. Once its controls have been learned, the system will become a handy tool for the smaller studio. The unit comprises a modified Commodore VIC-20 computer, and a companion interface box for linking a pair of audio and/or video transports, plus MIDI-equipped sequencers or synthesizers. Custom operating software is built into the system, to provide screen displays to timecode and MIDI status, plus locations of the various preset cue punch-in/out points.

The system provides autolocation and synchronization functions; operates all tape-machine transport controls plus punch-ins and -outs; generates non-drop SMPTE timecode; reads drop-frame, non-drop, 24 and 25 fps timecode; and allows the setting up of arbitrary timecode start times. The system can also generate 24-pulse sync, which in tests has proven to be the most accurate on the market. (The basic remote-control and autolocator SMPL System without synchronization capabilities generates 24-, 48-, and 96-pulse sync.) The Rehearse Mode allows you to check punch points, and provides an audible metronome signal for setting song tempo. The system also allows offsets to be established between master and slave machines, and will operate with either –10 dBv or +4 dBm equipment.

The SMPL System provides two modes of operation:
LOCK provides simultaneous control of two machines with remote-control access to the SMPL control box. Both transports are synchronized through the timecode recorded on each machine; this mode can be used for slaving together two multitracks, or for audio-video post work.
CHASE/SOLO provides remote control of the SLAVE machine only. This mode is useful in a situation such as ours — where there was no remote control access to the master VCR — and is aimed at those working with consumer videotape machines. The slave will follow start and stop commands given at the Master front panel.

The system really did a very nice job during the soundtrack recording,

and never failed to perform its assigned task. It allowed us to work quickly and efficiently and, by programming in the punch-lag compensation, we were able to obtain perfect punch-ins every time. Although punch points can be entered on the fly, by viewing the visuals, I really enjoyed the accuracy of programming in each point. The auto Search-to-Cue triggered after each punch-out point is also a nice convenience for instant review of the previous overdub.

My big problem with the system, however, lies not with its operation but in the documentation. The basic SMPL System is accompanied by a manual that is clear, well laid out, and has ample illustrations. In one sit-

ting, you could easily obtain complete mastery over the system's functions. The Chase Lock manual, on the other hand, was simply an addendum to the original. It was full of typographical errors, had no illustrations, and was more a description of system controls than an operational instruction. While the calibration procedure was easy to follow, and the interfacing guide also good, the Chase Lock manual only gave a vague idea of how to accomplish the desired tasks.

As an example, on power-up the basic SMPL System's sync pulse may be switched between three different configurations. However, this is not the case with Chase Lock (24-pulse is fixed), yet the manual makes no men-

tion that selection is not possible. Since there is no visual display of sync-pulse rate, we spent some frustrating time trying to synchronize the LinnDrum to the unit's output.

I have reviewed quite a few pieces of equipment for *R-e/p*, and consider myself to be a pretty good troubleshooter, but the SMPL System made me feel like I was missing something. The fact that three pro-audio dealers and a musician/electronics-genius friend of mine all had the same problems in deciphering the Chase Lock's operation helped restore faith in my technical abilities. Synchronous Technologies has assured me that, as of this writing, the formal manual is only "days away," and will probably be available as you read this article.

Southworth's Total Music turned out to be a mixed blessing for this session. There were several things Denis really liked about the sequencer software, some of which made his composing an easier task. In particular, he considered the program's editing capabilities to be the most extensive on the market. Once a sequence is recorded, any part of the music may be magnified several times, and note pitch, dynamics and duration

The Tascam Model 388 Studio 8 comprises an eight-input mixer and integral eight-track on quarter-inch tape deck with dbx noise reduction. Each input channel offers mike trim, three-band EQ, two effects sends, track routing and pan.

changed. Note values may be quantized, and attack and duration corrected.

If you have a hardware-based sequencer, such as the Roland MSQ-700, MIDI commands that has been combined onto one data "track" can be transferred to the Macintosh, and then separated by MIDI Channel for editing purposes. Also, undesirable

MIDI data, such as aftertouch, pitch bends, etc., can be filtered out during the recording of a sequence from an external keyboard. This feature of Total Music is handy for conserving sequencer memory, and for playing back the sequence on a synthesizer that doesn't recognize such data. All of these editing features help eliminate the need for re-recording the entire sequence when mistakes are made, changes must be incorporated, or unwanted MIDI data eliminated. Denis was particularly thankful for such versatility, since he often encounters directors who change their minds about a scene, and how the music needs to fit.

Total Music also features a special "window" display that enables the raw MIDI data stream to be viewed, and examined in greater detail. The entire program is a very "visual" experience, with information separated by MIDI channels and shaded according to dynamic values.

Although Total Music is extremely powerful and, we consider, offers the most features of any sequencer program on the market, Denis felt that it was complex and not particularly user-friendly. He wanted the power, but needed to have the freedom to create music without worrying about what the computer was doing, or which commands to give. Sequence chains were complex to set up, since the process involved copy and paste to a new sequence. There are 99 possible sequences, yet the only way to keep track of which ones are active is by scrolling through them one at a time. Only one sequence at a time may be looped, and data tracks and MIDI channels have no alphanumeric guide for instrument identification.

We encountered several program

The personal-use electronic-music studio used for *R-e/p*'s operational assessment.

bugs that contributed to Denis' opinion; the program crashed on him five or six times for strictly unknown reasons. We also spent a lot of time trying to figure out how to do "bulk data" dumps of voices, a procedure that worked only intermittently. After a phone call to Southworth Music, we were told that this was indeed a bug, and that they are working on the problem. Such bugs were really aggravating because, even though Southworth was aware that they existed, the company did not inform us of them until we called to ask about it. I do not feel that this is very responsible behavior, although, to be fair, Southworth had not had the opportunity to evaluate Total Music with the SMPL System.

Because Opcode Systems also claimed to provide Song Pointer capabilities, we tested its MidiMac Sequencer. As it turned out, MidiMac ran perfectly with the SMPL Chase Lock System. We were able to do all the punches knowing that the Song Pointer information would always "park" the synthesizers at the proper bar and note, waiting to receive their respective play command from the sequencer software.

The program turned out to be very user-friendly, and utilized three windows to handle all sequencing needs. It also was capable of accommodating 26 sequences of 16 data tracks each, which should be more that adequate for complex composing. For easy identification and setup, tracks could be given alphanumeric names and MIDI channel assignment. Any or all sequenced track may be looped at any time. The step mode is very easy to use with readouts of position by bar, beat and unit, and screen resolution may be changed at will. As Denis says, "Everything you need to keep track of is on the screen at all times. It does not have the extensive editing capabilities of Total Music, but runs faster and never crashed. It's a great, straight-ahead sequencer."

Well, there is no doubt that this session aged Denis and myself more than a little. We did spend a couple of frustrating days trying to figure out what we were doing wrong, only to

discover that the problem was actually due to the manufacturer. I feel very strongly about the policy of releasing products that have not been thoroughly Beta tested, and resent having to work through such problems while paying not only for the product itself, but also the studio time, musicians fees, etc. Our industry is too small for these types of things to go unnoticed, and I would like manufacturers to take notice that studio engineers and producers are *not* going to be tolerant of such a situation. (I wonder how long, for example, IBM would last if it released product with bad documentation and known bugs.)

Granted, there is a problem in the way different manufacturers view the

MIDI spec and sync codes, a [...] very expensive proposition to t[...] of the different equipment avail[...] today. I would hope, however, that th[...] people building this type of products, which we need to improve our recording and music making, take a more responsible approach to developing hardware and software that does not use the recording industry as a quality control and test laboratory. I am glad to report that several of the manufacturers whose products we reviewed here are now in communication with one another, and making sure that their products will interface properly in the future. Such communication is necessary if we are to further our musical endeavors. ∎∎∎

R MODEL ADR-68K DIGITAL REVERB FFECTS UNIT

The Model ADR-68K is said to perform the tasks normally performed by conventional outboard processors — reverb, sampling, special effects, and equalization — and features full MIDI implementation, including register recall and parameter control of reverb and effects functions.

Configured as a two-input/four-output design for "program splitting" — allowing two

independent programs to be run simultaneously — sound sampling is to 16-bit resoltion, with four discrete samples totalling eight seconds.

Each sound can be triggered independently or simultaneously (via MIDI-trigger inputs), while a remote-control unit offers touch-key display and slider controls, access to standard user-stored presets, and a "help" facility. In addition, a removable RAM cartridge doubles the parameter memory, providing the user with preset storage and portability of favorite effects programs.

The Aurora is software based, and the company says that new programs and sound will be generated on a continual basis. Suggested pro-user price of the Model ADR-68K is $3,995.

URSA MAJOR
For additional information circle #75

AUDRA INTERNATIONAL PMM-1 MINIATURE MICROPHONE

Features of the new small-size condenser microphone, which measures 0.6 by 0.3 by 0.01 inches, include an omnidirectional pickup pattern; 20 Hz to 20 kHz quoted fre-

quency response; no distortion up to 147 dB sound pressure level; application boundary-layer mike; operation from 12- to 48-volt phantom power (or optional battery adapter); compatibility with most wireless-mike transmitters; separate capsule and pre-amp; and a three-position bass roll-off switch for flat, 12 dB per octave below 160 Hz or 6 dB

per octave below 80 Hz.

Other specifications include an output impedence of 50 ohms, minimum load impedence of 200 ohms, XLR-3 male plug on pre-amp output, and a quoted output noise of 26 dB A-weighted.
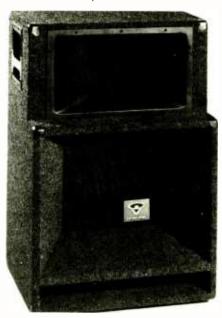
AUDRA INTERNATIONAL
For additional information circle #76

CERWIN VEGA MODEL V-31C AND V-37C SPEAKER SYSTEM

The V-31C is a two-way, full-range system, incorporating a 15-inch woofer plus high-frequency compression driver with horn, and is described as being ideal for high-performance keyboard applications. The 15-inch Model 153EV cone driver is front-horn loaded, and operates in a vented enclosure tuned to 15 Hz. A JMH-1 one-inch throat compression driver operates from the crossover point of 1 kHz, and utilizes an aluminum diaphragm with elastomer surround and edgwound aluminum wire voice-coil. The 90-by-40 degree horn coverage is said to provide optimum loading characteristics for the JMH-1 driver, with "wide, controlled dispersion over the entire operating bandwidth."

Preliminary specifications for the V-31C include a 50 Hz to 15 kHz frequency response; 150-watt power capacity; and an 8-ohm nominal impedence.

The Model V-37C features an 18-inch Model 188EB LF cone driver tuned to 40 Hz, and is similar in design and technical specifications to the Model V-31C, apart from an enclosed frequency response of 40 Hz to 15 kHz and EIA power capacity of 300W.

Both speaker models are finished in plywood and durable indoor/outdoor carpet, with built-in recessed handles, and are designed to integrate with other of the company's speaker components.

CERWIN VEGA
For additional information circle #77

CONSOLE AUTOMATION FOR APPLE II FROM SLICK SOFTWARE

Designed to run on a 128-Kbyte Apple II PC, the Slick Software software package provides console automation, studio inventory, and session scheduling via a tabular format.

The automation module, Studio Automation, features three menus: Session Information, Take Sheet and Mix Data — the latter illustrating module layout with track assignments, EQ selections, filter settings, cue sends, etc.

Studio Calendar allows entry of 1,000 items per file, and displays days, months, years, artist name, start and end times of sessions, and additional comments.

Studio Inventory lists data by category, serial number, name, purchase date, location and comments. All three software packages support printing options.

Suggested pro-user price of the three-module package is $895; individually, Studio Mixdown lists for $350, Studio Calendar $300, and Studio Inventory $300.

SLICK SOFTWARE
For additional information circle #78

CLICK TRACK AND STREAMER SYSTEM FROM OFF-BEAT SYSTEMS

Streamline is an IBM PC-based system that generates variable click tracks and visual clues (genlocked and keyed over VHS, Beta or ¾-inch video formats) for music scoring to film and video. An integrated hardware and software package, the system features timecode cueing capabilities, streamer and punch visual superimposed over a source video signal, and full storing of annotated cue information.

Audible clicks are outputted as standard line-levels, and can be used to trigger and control a Roland SBX-80, Garfield Electronics Dr. Click, or similar units. A timecode interface provides external cue triggering and regenerated timecode for re-recording.

The built-in Cue Editor enable control of timings, rate and beat frequencies in a spreadsheet environment. Sync points can be input and displayed as elapsed time, both drop and non-drop frame timecode, or feet and frames. Film click rates and standard metronome markings are both supported by the system, with a continuous range of intermediate settings.

Other features include automatically shaped accelerandos and ritardandos, free beats and warning streamers; bar structures with variable time signatures; click on/off anywhere in the cue; and visual cues entered by time bar and beat, or cumulative beats. Cues are summarized in a bar chart for on-screen reference or hard-copy printing, to show time signature, bar count, beat count, rate changes, elapsed time, and visual cue location.

OFF-BEAT SYSTEMS
For additional information circle #79

MANNY'S PROFESSIONAL AUDIO DIVISION

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New Products

PASSPORT MIDIPRO SEQUENCING SOFTWARE

As a new addition to the company's Master Tracks software, the new Midi Pro™ Series for the Apple II, IIe and IIc, plus Commodore 64 and 128 computers utilize a companion MIDI interface, and are claimed to offer more note storage than previous programs, as well as Real-Time, Step-Time, and Song Mode sequencing programs in a single software package.

The *Real Time* program provides access to all 16 MIDI channels, along with solo/mute functions on each track and "no limit to the amount of overdubs with the mix function." In addition, this program allows punch-in and -out editing with "trim" functions, and fast forward and rewind capabilities. Also, a new MIDI-Thru feature enables the user to monitor the output of the 16 MIDI channels from the keyboard controller, in addition to real-time recording of all controller, pitch bend, velocity, and aftertouch functions.

Step-Time utilizes the Quick Step Editor, which enables the user to step input and step edit pitch, duration, velocity, articulation, tempo, etc. Conversion of real-time sequences to and from step files for visual editing is also provided. Cut and copy/paste functions, and step punch-in and -out allows control over sequence editing.

Song Mode provides the assembly of songs using sequences created in the two previous programs. Up to 256 different sequences can be assembled using 256 different steps, and individual sequences within a song can be played back in any order, with tempo, transportation and repeat as desired. Individual tracks can loop independently, and solo/loop functions enable the use of tape sync with a "limited number of synthesizers."

Master Tracks has a suggested pro-user price of $249.

PASSPORT DESIGNS
For additional information circle #81

FOSTEX MODEL 260 COMBINED MIXER/CASSETTE RECORDER

The unit's six inputs comprise four switchable mike/line inputs plus two additional inputs that can function as tape returns, routing is to a stereo buss. Each input channel features a straight-line fader, mute button, gain-trim control, parametric EQ's, direct track assign or to the independent stereo buss, two auxiliary send controls, and monitor pan and gain controls.

The built-in cassette deck runs at 3.75 ips, and utilizes Dolby C noise reduction.

Other features include switchable LED bargraph meters, automatic monitor switching for punch-ins, automatic stop function with two-position memory, and top-panel patch points.

Suggested pro-user price of the Model 260, which replaces the Model 250, is $995.

FOSTEX CORPORATION
For additional information circle #82

AUDIO & DESIGN AD-MIX DIGITAL FADER

The new unit operates in conjunction with the company's Pro PCM701ES Section 2 conversion of the Sony PCM-701 digital processor, and provides adjustment of the digital signal level from infinity to either unity gain or +12 dB (switch option). If required, the unit can be used to combine an off-type digital signal with a second digital signal via the 701's A-to-D input section.

Other features include DC off-set nulling; 6 dB stereo balance; left or right phase reverse; and left/right channel reverse. Another useful feature, the company claims, is the ability to improve quantization noise of a Sony PCM-1610 by replaying the digital signal through Ad-Mix' dither circuitry via the Pro 701ES of 1610/30 interface. All Ad-Mix functions can be used with 1610/1630 units, in either send or return modes.

AUDIO & DESIGN CALREC
For additional information circle #83

AKG MODEL D70ME CARDIOID MICROPHONE

The new dynamic microphone has medium impedance of 200 ohms to 1 kohm. and equipped with a three-pin XLR connector, built-in pop screen, and Model SA-40 stand adaptor.

Specifications include a quoted 128-dB maximum sound pressure level, allowing close-up placement for instrument or amplifier miking. The mike's proximity effect is said to offer a "warmth and body" for vocal work.

AKG ASSOCIATES
For additional information circle #84

ASHLY CL-SERIES OF COMPRESSOR/LIMITERS

According to the company's VP of sales, Bob French, the CL50 and CL52 are single- and dual-channel compressor/limiters that are capable of being used, respectively, as two separate mono units, or as single stereo units. The CL52 features LED meters for both gain-reduction indication and output level, and utilizes a single rack space.

"The units provide tight control of peak levels without the usual side effects," says French. Other features include dual time constant release circuit; a soft-knee threshold detector; and separate attack, release and compression ratio controls.

ASHLY AUDIO

For additional information circle #86

SHURE MODEL FP32 STEREO PRODUCTION MIXER

Measuring just 2.25 by 7.25 by 5⅝-inches (H×W×D), the Model FP32 broadcast production console features two transformer-isolated outputs (one for each stereo channel); three transformer-isolated input channels (each containing level control); center-detented stereo pan pot; concentric stereo master gain controls; and the ability to switch between mike- or line-level. Also provided is a built-in 18 VDC phantom-power supply. Quarter-inch headphone jacks with level control, and a monitor input for monitoring an accompanying VCR are featured.

The FP32 includes a built-in slate microphone with automatic gain control, plus slate tones for identifying recording locations, a built-in limiter with adjustable threshold, dual VU meters, 12-VDC external power jack, stereo auxiliary-level outputs, and carrying case.

Suggested pro-user price of the Model FP32 stereo mixer is $1,200.

SHURE BROTHERS, INC.

For additional information circle #87

ORBAN MODEL 275A AUTOMATIC STEREO SYNTHESIZER

Designed for Stereo TV applications, the stereo in/out device incorporates two methods of automatic recognition — single-channel and mono-stereo — to determined when to synthesize stereo from a true mono input, and when to bypass stereo imaging.

Single-channel recognition mode monitors the absence of audio on one channel, thus crossfading to stereo synthesis from the remaining channel. Orban says that this method is recommended whenever it is possible for a TV station to route all mono audio solely via one channel.

Mono-stereo recognition mode, which is optional through a rear-panel switch, analyzes certain correlation characteristics of a two-channel input to decide when to initiate stereo synthesis, and automatically crossfade between stereo synthesis and bypass modes.

Other features include two synthesis modes (narrow and wide variations); single-ended noise reduction; polarity detection; fully balanced, +26 dBm inputs and outputs; recessed front-panel switches; and a user-control interface port for automated operation.

Suggested pro-user price for the Model 265A is $1,895; an optional remote control costs $295.

ORBAN ASSOCIATES

For additional information circle #88

New Products

ASHLY SG-33 STEREO NOISE GATE

Designed to control leakage and background noise in recording and live-sound environments, the SG-33 automatically atten-

uates signals that fall below a user-selected threshold, while passing audio that reaches or exceeds threshold.

Features include 60-dB threshold range; balanced or unbalanced inputs and outputs; stereo patching for tracking of two or more gates; front-panel in/out bypass switching for each channel; and control adjustments for threshold attack time, hold time, fade or release time, and noise floor.

ASHLY AUDIO
For additional information circle #91

MEMORY EXPANSION KITS FOR SEQUENTIAL PROPHET 2000 AND 2002

The Model 877 and 878 kits increase the onboard memory from the standard 256K to 512 Kbytes, doubling the maximum sampling time to a total of 32 seconds. The Model 877 is designed for any Prophet 2000 equipped with a single-sided disk drive, and carries a suggested retail price of $450, plus installation. The Model 878 memory upgrade is for units already equipped with double-sided drives, and for all 2002s; it has a SRP of $300, plus installation.

Depending on the sampling rate utilized, each of the Prophet's two memory placks can store samples having durations of up to 16 seconds (at 8 kHz), eight seconds (16 kHz), or six seconds (20 kHz).

SEQUENTIAL
For additional information circle #90

JBL LAUNCHES NEW RANGE OF STUDIO MONITORS

Designed to provide greater accuracy and tighter response tolerance, the new Model 4406, 4408, 4410, and 4412 studio monitors are said to deliver smooth frequency response to beyond 27 kHz, due to a perfected process of using pure titanium in a ribbed-dome tweeter. To ensure that the tweeter maintains rigidity under the extreme forces produced by high-amplitude, transients signals, a pattern of rids is formed into the dome of the tweeter to increase strength and stiffness. JBL's patented diamond surround configuration controls the second resonance, thereby enhancing the tweeter's high-end clarity.

While many manufacturers are said to

overlook the frequency-dividing network in studio monitors, JBL claims to have engineered its networks to allow optimum transitions between drivers in both amplitude and phase. Continuously variable, calibrated reference controls enable adjustment for room variations, specific equalization, or personal preferences.

The two-way Model 4406 monitor is

designed for console or close-in listening, and features a 6½-inch polypropylene woofer. The two-way Model 4408 was developed for broadcast applications, and contains an eight-inch fiber-cone woofer. Engineered with a vertical-line array to capture spatial detail at greater listening distances, the three-way Model 4410 has a five-inch midrange transducer, and a 10-inch laminate-construction woofer.

Also made with a laminate-woofer cone, the three-way Model 4412 features a tight clustering of components for close-in monitoring, and its 12-inch woofer has a three-inch diameter, ribbon-wire, copper voice coil for greater power handling and lower distortion.

JBL PROFESSIONAL
For additional information circle #94

TWO STEREO MONITORING SCOPES FROM B&B SYSTEMS

The AM-1B Phasescope is a stereo audio phase and level monitoring instrument intended for use primarily in video and broadcast post-production facilities. A CRT scope, plus VU and LED peak-reading displays, enables correct stereo phase information to be verified visually. Individual left and right audio channels, and timecode, can be selected for display.

The AM-2B Phasescope is designed with a smaller VU meters and CRT scope than the Model AM-1B, and may be used to check stereo phase and separation, verify peak levels for headroom availability, and measure average audio levels.

Suggested pro-user price of the Model AM-1B is $2,800, and $1,800 for the Model AM-2B.

B&B SYSTEMS
For additional information circle #92

CONEX ELECTRO-SYSTEMS MODEL AS-101 AUDIO SWITCHER

The new unit allows 10 stereo sources to be switched via microprocessor control to a stereo output. Illuminated front-panel buttons provide a visual indication of switcher status. Switching can be activated via a remote-control unit from up to 1,000 feet away; several remotes may be connected in parallel to the rack-mounted master unit.

Optional plug-in boards for the rack unit include RS232/422 interface for external computer manipulation of the switching designations; a relay board for external control of tape machines; and a system board that is said to provide a "simple automation system" when connected to a personal computer.

Output and input level controls are accessible from the front panel; all inputs can be individually switched for 10-kohm bridging or 600-ohm loads. Distortion is quoted at less that 0.01% THD, and signal-to-noise ratio –80 dB.

CONEX ELECTRO-SYSTEMS
For additional information circle #125

SHURE MODEL 587SB-7C MICROPHONE

Featuring a shock mount, high-density rubber support around the mike cartridge, plus a built-in spherical pop filter, the new

model is "perfectly suited to any application which requires a unidirectional microphone for recording, sound reinforcement, or broadcast applications," according to Sandy Schroeder, Shure's marketing manager.

Provided with a lockable on/off switch, swivel adaptor and platinum beige finish, the 587SB-7C lists for $90.

SHURE BROTHERS, INC.
For additional information circle #93

NEW MOS-FET POWER AMPLIFIERS FROM ASHLY

With power ratings of 225 watts RMS per channel for the Model FET-200, and 675 watts per channel for the FET-500, the two new amps feature balanced inputs, tamper-proof rear-panel controls, selector switches for mono and bridging operations, forced air cooling, and three-color LED output level meters.

ASHLY AUDIO

For additional information circle #124

CMX ANNOUNCES COMBINED TIMECODE EDITING AND CONSOLE AUTOMATION SYSTEM

Described as the first such system of its kind, the CASS-1 (Computer Aided Sound Sweetener) simultaneously controls up to six tape transports and 15 additional audio sources. Designed primarily for video post-production and audio-for-video facilities, the unit uses timecode references for level adjustments during a mix, with a reported offset accuracy of 333 microseconds (0.01 video frames).

The system will interface to most audio consoles equipped with VCA-controlled faders, CMX says, and features Recall and Modify functions, in addition to mono and stereo recording via CMX keyboard and Edit Decision List software.

All storage and retrieval of automation data is via floppy disk, with archival storage to and from an internal hard-disk drive. The system has a memory capacity for up to 16 faders in preview or record mode. Dynamic graphic displays on a CRT show current fader levels at all times.
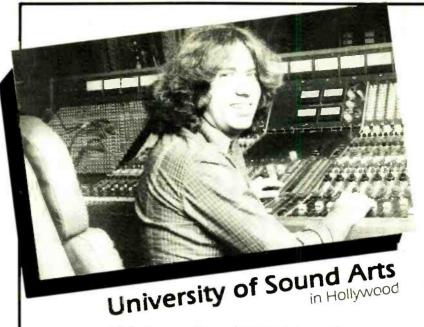
CMX CORPORATION

For additional information circle #95

ATUS™ MIKE STANDS AND BOOMS FROM AUDIO-TECHNICA

The microphone stands and related products make up three categories: floor stands, boom assemblies, and combinations (floor stands with booms). Since the new products are made to A-T specifications by a West German manufacuturer, they will be marketed under the name ATUS, rather than Audio-Technica. Even so, according to A-T national sales manager Mark Taylor, they will be distributed exclusively by Audio-Technica, and are said to offer design features not offered in any other line.

The two floor stands extend from 35 to 65 inches in height; the Model AS500-C is

chrome plated, while the AS500-B matte black. Bases of both units are die-cast metal with fold-out legs. Stand uprights are locked in place by a turning a knurled knob that pushes a concave pressure plate firmly against a plastic sleeve to provide a secure locking action with lower torque.

The two ATUS boom assemblies extend from 16½ to 32 inches; Model AB500-C is chrome plated, while the AB500-B is finished in matte black. The unit's angle adjustment device, it is claimed, is the most advanced design available, permitting easy, reliable settings.

The two floor stand/boom combinations include the Model ASB450-C, a full-size (35 to 62½ inches), chrome-plated stand with a 33-inch, single-section boom; and the Model ASB510-B, a low-profile stand (12½ to 24½ inches), finished in matte black, and suggested for use with drum miking plus other low-level applications.

AUDIO-TECHNICA U.S., INC.

For additional information circle #98

NEVE INTRODUCES V-SERIES CONSOLES

The V-Series is derived from the 51-Series range, which were designed primarily for broadcast and video post-production work. The new console line is intended for use in

broadcast multitrack teleproduction, film and video audio post-production, and music recording, and features optional 60-, 48-, or 36-buss configurations, containing selected standard features such as individual channel dynamics (limiter, compressor, gate) with external keying; separate multitrack and mixdown signal paths with individual faders; and the company's four-band Formant Spectrum Equalizer.

Other capabilities include in-line or separate monitoring and solo system with selectable reverb and effects returns, and all electronically balanced inputs.

RUPERT NEVE, INC.

For additional information circle #99

NEW DIGITAL SAMPLING PIANO FROM ENSONIQ

The 16-channel, MIDI-capable piano features 12 preset sounds, including bright and mellow electric piano; two marimbas; electric bass; upright bass waveforms; grand piano; vibes; percussion; and mallet. Acoustic piano variations are bright and mellow timbres, and honky tonk.

The unit features 76 piano-size weighted keys, giving the player control over dynamics and timbre with touch sensitivity, sustain, and sostenuto pedals — the latter two being MIDI capable. A transpose switch allows transposition of sounds up a major 4th and down a diminished 5th.

Built-in stereo chorus and stereo headphone jack are available, while channels for bass sound, and a separate LF output allows the user to send signals to a separate amp.

ENSONIQ

For additional information circle #100

NEW DIGITAL SEQUENCER FROM POLYPHONIC FX

The 16-bit system is a stand-alone sampling system that directly references SMPTE time-code locations for storage and replay of sound effects and dialog elements. Utilizing a 40-Mbyte internal hard-disk drive, 8-MHz internal clock, and 640 Kbytes of RAM, the unit is designed as a 16-channel software-based system — for simultaneous or sequential playback — with standard line-level outputs.

The unit can reproduce up to six voices per effect from digitally recorded sounds previously stored on hard disk. A 250-Mbyte optical Laserdisc unit is offered for optional storage, and which can also serve as a central sound library for local-area networking of

more than one system.

As a "completely integrated hardware computer system," the unit is reported to provide a cost-effective solution for frame-accurate playback of sound effects to SMPTE timecode; storage of effects in order of occurance; and automatic load of each effect prior to timecode location. The system is expandable to 256-channel playback (16 banks of 16 channels per bank).

Suggested pro-user price for the Polyphonic FX System starts at $16,000, including all software and hardware.

POLYPHONIC FX SYSTEMS

For additional information circle #101

NEW RANGE OF ELECTRONIC CROSSOVERS FROM ASHLY AUDIO

Constructed in two-, three-, and four-way models (available in both mono or stereo formats), the XR-Series feature 2 or 18 dB per octave slopes, and slate-variable filters. Both high- and low-pass filter functions can be performed with one filter section, and two or three filters can be cascaded for three- or four-way crossover networks. In addition, crossover points are individually tunable over a five-octave range, and the company's crossover damping point is adjustable. This latter feature functions as an equalizer band centered at the crossover point, to allow flat summed response with either slope.

Special output stages with feedback level controls provide up to 20 dB of gain. An "electronic power monitor" controls output stages, preventing turn-on transients without the use of relays. A peak overload circuit, with LED indicator, monitors all critical signal points in the crossover, and illuminates when signals are within a few dB of clipping.

ASHLY AUDIO

For additional information circle #102

The *R-e/p*
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of Cutting and
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no direct correlation between out-of-band and in-band signal handling capability. (While I won't deny that the picture-perfect squarewave response [with ideal Gibb's phenomenon squiggles] encouraged me, I was more influenced by the $169 price tag on my new Compact Disc player!)

On the subject of CD players, and how they sound, several friends of mine have reported dramatic differences between various CD players that I can't explain from measured performance. Another friend, who has a part-time job repairing hifi gear, reports that he has repaired an inordinate number of CD players because their lasers were out of alignment. It's probably safe to assume that some much larger number of players will be out of adjustment, but within the pull-in range of the error-correction circuitry. Depending upon their degree of misalignment, such players could sound subtly inferior to properly aligned units. On the other hand, my friends could be hearing things.

About Sinewaves

I've argued that the squarewave can easily be too *rigorous* for meaningful measurements; on the other hand, the simple sinewave can sometimes be too *easy*. Of course, this wasn't always the case and, to keep things in perspective, our expectations have increased along with the near disappearance of THD from modern equipment. We should resist the temptation to revert to out-of-band signals, just because our equipment measures so well. In fact, even simple sinewaves above 6 to 10 kHz are suspect, since their overtones are out-of-band.

The ideal test signals will always be music or voice, but they are difficult to standardize and quantify. What good are tests without repeatable numbers?

Thanks to the advent of digital recording, we are swiftly approaching the point where we can generate reference or standard music and vocal tracks. We would be starting all over again from ground zero regarding the significance of peak versus average deviations from this standard, but, once sorted out, the results would be eminently meaningful. Until then, there are some sinewave-based tests that are very useful. Combining two or, better yet, three (after Cordell) high-frequency yet still in-band sinewaves, will generate a higher slew rate than either signal alone, while still being undeniably in-band. Furthermore, this waveform will generate intermodulation distortion products at some lower, also in band, frequency. (You don't have to worry about input or output filters erasing their own distortion products from your measurements!)

CONCLUSION: In the absence of definitive vocal or music reference tracks, and a way to quantify a piece of equipment's deviation from same, we must rely upon conventional test signals. Use judgement and caution when applying stressful, high slew rate tests, since their result may not be meaningful. Whenever possible, try to match the test to the equipment. ∎∎∎

MIDI EQUIPMENT UPDATE

New Products Unveiled at the Winter NAMM Exhibition, Anaheim

by Bobby Nathan, Unique Recording, New York

MIDI was once again the buzz word at the Winter NAMM Show, held during early January in Anaheim, CA. Many leading manufacturers showed components that could be sold as independent units, or as a complete "MIDI System" comprising a MIDI-equipped drum machine, sequencer and synthesizer.

Yamaha introduced its answer to the need for a low-priced MIDI-equipped system, and which incorporates a choice of either the **DX-100/DX-27** synthesizers, **RX-21/RX-21L** drum machines, and the **QX-21** master sequencer. The DX-100 and DX-27 were conceived, Yamaha says, for the entry-level keyboardist/synthesist; the 100 features a four-operator, FM generation system, and a 49-note keyboard. A total of 192 ROM preset patches are included, as well as 24 user patch memories. Pitch-bend modulation and breath-control functions are also supported. Although the DX-100 does not feature a velocity-sensitive keyboard, velocity functions can be controlled through external MIDI connection. Weighing less than six pounds, the 100 can be battery operated for use at the beach. The DX-27 is provided with basically identical features as the 100, but with a full-keysize, five-octave keyboard.

The other two basic components of the Yamaha MIDI System are the RX-21 and RX-21L drum machines (the latter a latin percussion version), and the QX-21, a lower-priced version of the QX-7 sequencer. Note memory has been expanded from 6,000 to 8,100 notes, and real- and step-time quantization extended to a 1/64th-note increment.

Roland also showed its new low-priced MIDI System components, including the **Alpha Juno 1 and 2** synthesizers. The designation "Alpha" is said to come from the unit's one-knob editing capability via an Alpha — video-style editing — control, and 32-character liquid-crystal displays. Other features include the ability to sense velocity and aftertouch through MIDI; 64 patches in internal memory; load/save to standard cassette tape; one DCO (digitally controlled oscillator) per voice, with 14 basic waveforms; and an eight-stage envelope. The Juno 2 is identical to the Juno 1, except for the addition of a five-octave, velocity- and pressure-sensitive keyboard, and the ability to store sounds on an optional M64C RAM cartridge.

Also to be seen: the low-priced **TR-505** MIDI drum machine, which includes 16 PCM digital sounds; 48 preset plus 48 programmable rhythm patterns; rhythm chains up to 423 measures; control of sounds via external audio triggers; AC/DC operation; and complete MIDI In/Out/Thru and MIDI Clock Sync.

Taking off where the MSQ-700 left off —

for the same price — the new **Roland MC-500** MIDI sequencer features four data tracks (each holding a total of 16 MIDI channels); a separate rhythm track; track merging; individual MIDI-channel editing of merged tracks; and a 20- by two-character display. Data storage capacity is up to 30,000 notes in internal memory, or up to 100,000 notes on a built-in 3.5-inch disk drive.

Thinking of purchasing a sampling synthesizer? Well, this year finding an answer to the question of "Which one to choose?" is going to be even harder, beause all the major manufacturers showing at NAMM had what seems to be "Sampling Fever."

The new **Korg DSS-1** 12-bit/32-kHz sampler allows up to 16 sample-split points to be set on the unit's five-octave, velocity- and pressure-sensitive keyboard, and comes complete with a built-in 3.5-inch disk drive. The DSS-1 can be considered truly unique with its twin digital delays on each of the stereo outputs. Complete with VCF, VCA and LFOs, once a sound is sampled into the unit it can easily be

edited. Looping and truncation features are also included.

Roland announced the new **S-10** and **S-50** sampling keyboards, the latter comprising the first low-priced unit to offer 16 voices with 16 oscillators, VCFs, VCAs, LFOs, and 48 envelope generators (for changing oscillator pitch over time). Sampling time is fixed at 17.2 seconds; while the sampling frequency is set nominally at 32 kHz, it can be altered to 15 or 7.5 kHz. Other features include automatic and manual looping; one-shot and alternate (backwards/forwards) looping; a built-in 3.5-inch disk drive; integral on-screen software for editing waveforms, loops, truncation, envelopes; and built-in RGB and NTSC monitor interfaces, thus doing away with the requirement for an outboard computer.

The S-50's keyboard has the same features as Roland's Master MIDI keyboards (MKB-100, 200 and 300) with split MIDI channels, velocity and touch sensing, etc. Sounds can be sampled on all 61 keys, and then split and doubled. A Stack mode allows four separate samples to be layered on one key, while Scratch mode enables a sample to be played like a DJ for you Dubbers of Doom!

The smaller S-10 incorporates a 49-note keyboard, four sampled splits, and a built-in 2.8-inch "quick-disk" drive. It offers four seconds of sampling at the preset 32-kHz sampling frequency, and also has onboard software similar to the S-50.

Those *R-e-p* readers that heard the **Roland MKS-20** Digital Piano rackmounted module last October at the New

— Yamaha SPX-90 Effects Processor —

— Roland MKS-20 Digital Piano Module —

York AES Convention probably found it hard to believe that the sound was being synthesized. While the company's new technology, known as Structured/Adaptive Synthesis, sounds like samples, it was really cooked up from the waveform cookbook! The eight available sounds include variations of Bosendorfer, Steinway, Yamaha and Baldwin pianos, plus Fender Rhodes and Vibes. Also on show: the RD-1000, a complete 88-note, wooden-weighted keyboard with a built-in MKS-20 sequencer and full MIDI In/Out/Thru.

Sequential introduced the **Prophet 2002** rack-mounted sampler, which is identical to the existing Prophet 2000 apart from the following added features: a double-sided, 3.5-inch floppy disk drive (also capable of reading single-sided disks made on earlier Prophet 2000s); and optional RAM capacity increased from 128 to 256 Kbytes, thereby providing a total of 16 seconds of sampling time.

Akai introduced the **S-900**, a high-end 12-bit/eight-voice, MIDI-equipped polyphonic multisampler with 32 sample points. The unit offers a 40-kHz sampling frequency, and 12 seconds of sampling time. Eight individual trigger inputs and

outputs allow the device to be used as a sampling electronic drum head! Built-in VCF, VCA, loop crossfacing and storage via an integral 3.5-inch disk drive also are featured.

Fairlight was showing several new features for the CMI Series III, including 16 voice/channel cards (expandable to 90); 50-kHz stereo sampling frequency (100 kHz in mono mode); up to 14 Mbytes of waveform RAM (allowing two minutes of 50-kHz sampling time); a 140-Mbyte Winchester hard disk; a 16-track, 16-voice per track Page R sequencer; and the company's new programming system — CAPS.

As for new synthesizer technology, several U.S. manufacturers led the way. **Ensoniq** introduced its new **ESQ-1**, a 61-note/eight-voice synth equipped with 40 internal patches; three different waveform oscillators per voice (with over 32 multisampled and synthetic waveforms); plus a built-in eight-track MIDI sequencer with a 2,400-note capacity and cartridge slot for storing patches. The company also debuted the new **ESP-1** digital piano, which features a 76-note keyboard with 12 multisampled digital voices, including grand and electric pianos, marimbas, and upright

bass.

Sequential was showing its **Prophet V/S** eight-voice, 60-note velocity/pressure sensitive digital vector synthesizer, which reminds this writer very much of the PPG 2.3. Each of the unit's eight voices is built from four, 12-bit oscillators, allowing each oscillator to be controlled independently using any of the 128 complex waveforms, including white noise. Access to over 200 program patches is provided via an external RAM cartridge.

The **Roland Super JX-10** is an enhanced version of the JX-8P that features a 12-voice, velocity- and pressure-sensitive keyboard; split and laying capabilities; 50 preset and 50 programmable patches; 64 patch preset memories; a 170-note sequencer; and a new function called "chase play" that allows the firing of two voices as a delay function. The device also incorporates the Alpha data-entry controller with 32-character display, and a 650-step real-time sequencer with loop switching, overdub and punch-in capabilities.

The selection of new MIDI sequencer software on show at the Winter NAMM Show was staggering. Designed to run on an Apple Macintosh, **Mark of the Uni-**

corn's **Performer** features 200-plus tracks (each with separate or multiple MIDI channel assignment); a 50,000-note capacity; numeric MIDI Note/Event editing; and 480 clock (ticks) per quarter-note resolution. Screen layout very much resembles that of a tape machine, with play, record, auto record (punch-in and out), stop, fast-forward and reverse functions. The package is said to work with all existing MIDI interfaces designed for the Macintosh.

Musicworks introduced the **Megatrack XL-1**, an updated version of its existing software for the Mac. Speed is said to be the key word for this new package, along with unlimited track capability, and extensive graphic note editing with zoom features. In addition, two tracks can be viewed simultaneously, with bar numbers in sync, and notes cut and pasted from one track to another.

Opcode released version **V2.0** of its **Midimac Sequencer**, which now features 26 sequences with 16 tracks per sequence, step editing, simultaneous recording on multiple MIDI channels, and MIDI event filtering.

Mimetics was the first company to show low-cost software for the Amiga personal computer. **Soundscape** is a modular package, and can be purchased to handle just MIDI sequencing or sampling, or both. Features include unlimited tracks, event editing, cut and paste, etc.

Sight and Sounds' Midi Ensemble sequencer is designed to run on an IBM PC equipped with a minimum of 256 Kbytes of RAM, and a Roland MPU-401 MIDI interface. The software package features 250 tracks, automated punch-in/out, programmable tempo change, programmable metronome, solo/mute on any data track, transposition of any track, and programmable volume for each track (the latter for MIDI-equipped instruments that read such data). Three modules make up the package: Recorder, Event Editor (graphic note editing), and Phrase Editor (which allows autocorrect, merge, replace, insert and delete of previously recorded tracks). Future modules will include Manuscript (for music printing), and Arranger, a series of musical arrangement and drawing board units.

Digidesign, the company that developed **Soundesigner** software for the Mac, introduced new versions for the Ensoniq Mirage and Sequential Prophet 2000 samplers. Like the original version for the E-mu Systems Emulator II, the new software features display of three sounds simultaneously, and sound editing with full cut and paste functions for re-arranging sounds within an accuracy of approximately 33 microseconds. Other features include the ability to draw waveforms from scratch, or the repair of sampled sounds; full digital equalization functions; and three-dimensional waveform analysis. Having been transferred to the Mac, sampled sounds downloaded from an EII, Mirage or 2000 can be uploaded into any of the other devices. [And possibly open up the way towards a universal sound-file format for synthesizers from different manufacturers — *Editor*.]

Key Clique introduced a new series of custom cartridge ROMs for the Yamaha

...continued overleaf—

MIDI UPDATE

DX-7, with 64 presets on each cartridge programmed by factory programmer Bo Tomlyn. The 16 cartridges currently available are titled as follows: Top Forty, Orchestral, Splits, Studio, Stereo, Sound Efx, Hard Rock, Electro Techno, New Wave, Analog, Church, Wedding Band, Latin, India, Easy Listening, Country/Western, and Unique! Also from Key Clique: the **Sys Ex** software for Commodore, Apple IIe and IBM PCs that allows quick data storage of sound patches, sequencers and drum machine pattern data via MIDI. The list of synths, sequencers, and drum machines supported is quite elaborate.

Club Midi showed a Patch Librarian for the IBM PC, entitled **the Midi**. The low-cost package supports patches for the Yamaha DX/TX; Oberheim OB-8, Expander, Matrix, 10 and 12; Korg SW6000 and DW8000; Roland JX-8P and MKS-80; and Sequential Prophet 5, T8 and 600. Included in the listing of patch titles is a "Remarks" section for noting patch comments for mod-wheel usage, pitch bend, etc.

Storing Emulator II samples got you down? **Optical Media Services** introduced the **CDS3 CD-ROM** and inter-Sounds library that represents the equivalent of 536 Emulator 5.25-inch floppy disks. The DC-ROM disk has a total capacity of 1,106 EII floppies, or around 550 Mbytes of data. The CDS3-Mac package includes software for the Apple Macintosh, and is said to be totally compatible with Digidesign's Soundesigner software.

For the Apple IIe, **Passport** introduced the **Master Tracks** sequencing package, which features unlimited tracks, step-time recording, and Store-a-Separate-Song mode for assembling songs with up to 250 steps; punch-in/out in real time; MIDI Thru while recording; and 8,000 MIDI event memory. File structure is totally compatible with Midi4 Plus and -8 Plus Polywriter and Music Shop.

Linn Electronics introduced **Midi-studio**, a new system that is essentially a rack-mounted Linn 9000 with a remote lap pad to program drum patterns and sequencer setups. The unit comes complete with samling software (variable from 10 to 50 kHz — 10 seconds total capacity at 30 kHz — divided among the 16 drum pads); and a built-in 3.5-inch disk for storing drum patterns, sound samples and sequencer patterns. Included with the Midi-studio are 50 sound samples.

Other new MIDI innovations included the **Yamaha SPX-90** multi-effects signal processor with MIDI control. Included are several reverb and early reflection effects, digital delays, doubling, pitch and amplitude modulation, pitch change, compression, noise gating, parametric EQ and even auto-panning effects. As well as the 30 factory presets, 60 user-programmable patch variations are provided. What's perhaps most impressive about the SPX-90 is the unit's ridiculously high-sound quality versus its low cost. Yamaha also introduced the **MEP-4** MIDI event processor that accepts and modifies incoming MIDI data prior to output.

J.L. Cooper introduced a number of new MIDI innovations, including **MIDI Link**, a rack-mounted MIDI Program

Manager that allows a master MIDI controller or sequencer to send different MIDI program patch numbers to individual MIDI slave devices. Features include up to 99 programs, and a one-in/six-out format. The company also introduced **Midi Mute**, an eight-channel MIDI-equipped muting device for recording or live keyboard mixing. When connected between the instruments (or effects) and mixer inputs, the unit receives and generates MIDI Note On/Off commands, and converts them to switch closures via internal relays. In this way, a MIDI sequencer can be used to run a mixer channel on and off automatically. The unit is expandable in groups of eight channels, and can easily control 24 channels of a recording/live-sound console.

MIDI input mixing? Although they said it couldn't be done, **Garfield Electronics** has come up with a fix. **Traffic Controller** allows four MIDI Ins to be mixed simultaneously to four MIDI Outs — great for having more than one master MIDI keyboard hooked up, and also a computer/editor/librarian! **Kamlet** also introduced **Midi Merger**, which enables two MIDI Ins to be mixed to a single MIDI Output.

And how about MIDI-controlled Automation? **Akai** showed an eight-channel version that memorizes fader level, panning, three-band EQ, echo and cue sends and auto-fade in and out. The mixer is equipped with mike and line inputs, and 99 MIDI preset patches that could control changes in level, panning, EQ, etc. from any master MIDI keyboard. (This board would be great for on-stage use!)

MIDI guitar anyone? **K-Muse** introduced the **Photon** laser pickup system that looks, at long last, to be the guitarist's answer. The system incorporates 16-bit technology, and its patented laser pickup connects to a high-speed pitch converter/controller unit that takes care of MIDI channel, patch changes, velocity, etc. Pitch bend (via strong bend) velocity (via pick/string attack) is transmitted with a trigger delay of only three milliseconds! There will also be a "hyperspeed" version complete with a guitar that uses only high-E strings to make pitch conversion even faster. A version for electric bass is also planned in the near future.

Several software updates were also announced, most of which added new features and fixed previous bugs. The following are a few of the software updates released at NAMM:
• Linn 9000 version 5.x software is a free update that fixes the bugs previously found in the unit;
• Southworth's Total Music V1.0 is a free update to all present owners;
• A MIDI upgrade for the Oberheim DMX drum machine that adds MIDI In/Out and Clock functions at a nominal charge;
• Yamaha's QX-1 update and software upgrade includes several new features for increased speed, and comes complete with a new panel overlay for added and changed job commands, plus a new owner's manual; and
• Yamaha also announced a back-lit LCD option for the DX-7, making it easy to view parameters in low-light situations. (Both of these latter updates are also available at a nominal charge.)
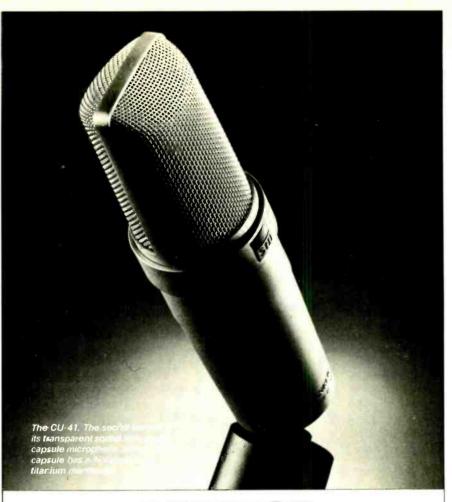
■■■

News

marketing background; Dunkle is a sales rep with several years of experience in Florida, and Collins is an audio systems designer/engineer.

For more details contact: ESM, Inc., 4405 Vineland Road, Suite C-8, Orlando, FL 32811. (305) 423-2363.

DIGITAL AUDIO DISC CORP. TARGETS FOUR MILLION MONTHLY PRESSING BY 1987

Digital Audio Disc Corporation, the first Compact Disc manufacturing facility in the United States, plans to increase pressing capacity by 200% over the next two years. As it commemorated the pressing of its 10 millionth CD, the facility expected to double current manufacturing capacity to two million discs per month by the end of 1986, and again double capacity in late 1987. The Terra Haute, ID, facility is a wholly owned subsidiary of Sony Corporation of America and presented the 10 millionth CD to Norio Ohga, president, Sony Corporation.

Digital Audio Disc Corporation opened 18 months ago with a production capacity of 300,000 discs per month, and fewer than 100 employees. Since that time, both the manufacturing capacity and work force have more than tripled.

FIRST ANNUAL NARAS SAN FRANCISCO MUSIC FAIR

The San Francisco Chapter of the National Academy of Performing Arts & Sciences (NARAS) has announced plans for the first annual San Francisco Music Fair, to be held on June 27 thru 29 at the Concourse at Showplace Square. Presented by NARAS/SF and co-produced by Events West, the three-day gathering will feature pro-audio displays, educational seminars, ongoing live performances on two stages, plus demonstrations of audio and video equipment, musical instruments and accessories.

Anyone interested in participating should contact Kelly MacRae at Events West; (415) 383-9378.

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— News Notes —

• **The Los Angeles Chapter** of the **National Academy of Recording Arts & Science** has awarded a $600 "educational grant" to **Dick Grove Music School** student **David Goldblatt.** "This is only one of six educational grants we give each year," notes Ron Kramer, local NARAS Chapter president. "We also provide scholarships and grants to talented students at UCLA, USC, and Los Angeles Valley College, as well as the proceeds of major fund raisers for special music programs run by the Los Angeles Unified School District, and the Los Angeles High School for the Performing Arts." Prior to beginning his current course of study in composition, arranging and film scoring, Goldblatt studied at both Berklee School of Music and at Northern Illinois University.

• Due to "rapid expansion and unprecedented sales," **Renkus-Heinz, Inc.** has moved from its 6,000-square-foot facility to a new location with potential usable space of 30,000 square feet. The new site is on the corner of Armstrong and Langley, in Irvine, CA, within blocks of the old facility. While telephone and telex numbers remain the same, the new address is: 17191 Armstrong Avenue, Irvine, CA 92714.

• **Dolby Laboratories** has moved its corporate headquarters from Sansome Street to Portero Avenue. In the 10 years since Dolby relocated its headquarters from New York to San Francisco, the company is said to have enjoyed substantial growth in its three major areas: the manufacture and sale of specialized electronic equipment for signal processing and noise reduction; the licensing of Dolby Stereo films and sales of cinema processors; and the licensing of consumer audio processes, such Dolby B-type noise reduction. The new 70,000 square foor building more than doubles the work space for Dolby Laboratories' engineering, licensing, U.S. marketing and administrative staff. The company's new address is: 100 Portero Avenue, San Francisco, CA 94103-4813. ∎∎∎

— People on the Move —

• **B. Morgan Martin** has been appointed product manager for the SoundDroid Digital Audio Processing Systems at **The Droid Works.** His prime responsibilities will be to oversee the introduction of the SoundDroid product line, and supervise the installation of the first SoundDroid systems scheduled for delivery in mid-1986 at various facilities across the U.S. Prior to joining The Droid Works, Martin was western regional manager for Rupert Neve, Inc.

• **Dr. Roger Lagadec** has been named general manager, Technical Management, Communications Products Group at **Sony Corporation,** where he will be responsible for forming engineering strategy, developing industry relationships, standarization, and direct research and development in professional audio, video, and related products. After working for Sony in Europe for a period of three months, Lagadec will be relocated to Japan in May. For the past six years, Lagadec was product manager, Digital Audio, at Willi Studer AG, Switzerland, where he was responsible for several of the company's inventions and patents in digital audio. Lagadec received his PhD in digital signal processing and communications from the Ecole Polytechnique in 1975, and subsequently served there as an assistant professor of applied physics between 1975 and 1979. His research interests have included semiconductor physics, digital filters, fiber optics, data transmission and digital signal processing. ∎∎∎