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

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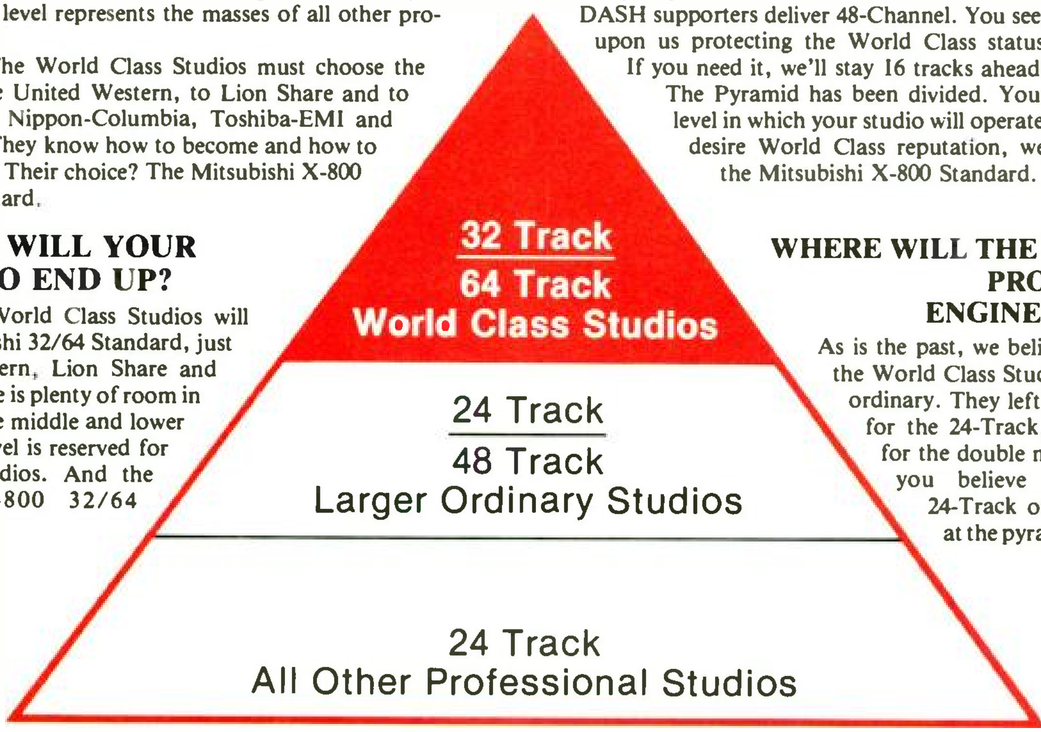

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AUDIO MYTHOLOGY QUESTIONNAIRE

from: **Dr. Eugene Zaustinsky
Department of Mathematics
State University of New York,
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I very much enjoyed John Roberts' excellent column on speaker leads, published in the October 1983 issue of *R-e/p*, and highly applaud his plans to go on to treat the questions raised in the questionnaire. Although I have no doubt that John will do a comparably excellent job when he treats the three related Questions 8, 21, and 24 [dealing with dB per octave crossover rates, audibility of phase shift and/or absolute polarity, and audibility of speaker phase coherence/time alignment — *Ed.*], these are much more difficult to treat if one is contrained in the amount of mathematics one can use. My recent paper[1] might prove helpful, since it discusses several relevant points which, to my knowledge, have not previously appeared in print.

I have been particularly interested in Questions 8 and 24, and hope that *R-e/p* readers will find the following additional remarks helpful.

Before proceeding to specifics, it is important to have in mind the fundamental importance of flat amplitude response. A very large part of the perceived subjective excellence of a loudspeaker is due to the flatness of its amplitude response and, for a stereo pair, accurate matching of the two individual speakers, as well as their individual responses.

To treat Question 8, it is useful to start by distinguishing between *electrical* and *electro-acoustic* (henceforth called *acoustic*, for short) crossovers. Electrical crossovers are characterized by their electric signal to electric output transfer functions (i.e., responses). The most familiar examples of these are the off-the-shelf active (electronic) crossovers, consisting of a complementary pair of Butterworth (6 or 18 dB per octave) or Linkwitz-Riley (12 or 24 dB) high-and lowpass filters, *both tuned to the same frequency*, which are commonly used in sound reinforcement applications. When the amplitude and phase responses of the individual drivers and the horizontal physical displacement of their acoustic centers are taken into account, these will not, in general, yield an acceptably flat amplitude of on-axis sum of acoustic out-put response. Bullock[2] shows that the best hope for a fortuitous acceptable result lies with fourth-order Linkwitz-Riley. However, the completely unacceptable result obtained in this way, shown in Figure 9 of my AES Preprint, shows that this electrical crossover is no panacea. Furthermore, with some driver pairs, better results are obtained with other crossovers, most commonly the 18 dB.

Apart from the above-mentioned fact concerning which of these electrical crossovers has the best prospect of producing a fortuitous acceptable result, it makes no sense to try to compare the four types (actually, there are six, as the 6 and 18 dB may be connected with either polarity). With a given pair of drivers, these crossovers will produce six different amplitude responses, all sounding different and none of which is very good. The reason that there is no agreement as to which sounds best is, quite simply, due to the fact that there is also no agreement as to which pair of drivers to use.

An acoustic crossover is characterized in terms of the signal-to-acoustic pressure output transfer functions (i.e., responses) it yields, when used with the *specific set of drivers* for which it has been designed. These transfer functions may be any of the four considered above, and there are also many other possibilities. The most familiar examples of acoustic crossovers are the fourth-order Linkwitz-Riley crossovers used in the well-known B&W 801 and KEF 105.2 loudspeakers, and the third-order Butterworth crossovers used in some other B&W products. There are, of course, many others. However, I mention these because further interesting and useful particulars, concerning these designs have been published in the technical literature.[3 4]

Since acoustic crossovers produce flat sum of on-axis acoustic output responses, they can be meaningfully compared. In doing this, however, because of the extreme sensitivity of the ear to very small amplitude response variations, it is almost impossible to insure that one is really hearing crossover differences, rather than differences in the accuracy to which the crossovers have been realized. The signal to on-axis sum of acoustic output's transfer function, in the cases considered, is that of a first- or second-order all-pass filter. Thus, one way to approach the question of the audible subjective significance of crossover order is to audition the equivalent all-pass filter with earphones.

In reference #5 Lipshitz *et al.* report the results of such experiments. With suitable synthetic signals, the filters corresponding to crossovers, with crossover frequency in the range from 100 Hz to 1 kHz, are all audible; at higher crossover frequencies, all appear to be innocuous. We conclude from this that, insofar as transient distortion is concerned, perceived differences between loudspeakers cannot be attributed to crossover *order*.

On the other hand, loudspeakers with odd- and even-order crossovers sound different because of the different acoustic radiation patterns they produce. Those who believe in the great importance of radiation pattern use fourth-order crossovers, and frequently refer to Linkwitz[6] for the proof that they are doing the right thing.

This argument cannot be easily dismissed. For one thing, there are two very fine commercial products, among the world's best, already mentioned, which use this approach. However, there are also several other loudspeakers, in the same class, which use odd-order networks. The actual trade-off is this: odd-order networks can yield both flat amplitude and flat power responses, while even-order networks may produce a power response dip at crossover frequency. It is my opinion that the audible significance of this difference has not been experimentally demonstrated. It is, in fact, conceivable that one type may be preferable in some rooms with some program material, and conversely.

It is my opinion that there is no such thing as a good, general-purpose off-the-shelf crossover, either active or passive, *of any one of these given fixed types*. All of the very best crossovers are acoustic, designed for a specific set of drivers on the basis of precision driver measurements. Concerning finished speaker systems, world-class examples exist, using all four orders considered. Of course, since all of these are acoustic, none in any way resembles a complementary pair of high- and lowpass filters tuned to the same frequency. After considering the next Question, we shall return to this point and consider the possibility of flexible, tuneable crossovers.

Turning next to Question 24, the expressions "time-alignment" and "phase-coherence" are loosely used, with a variety of meanings. [It should be pointed out that Time Align™ is a trademark of E.M. Long Associates — *Ed.*] A few people use "phase-coherence" to mean in-phase acoustic outputs, such as are produced by a fourth-order Linkwitz-Riley acoustic crossover, which we have already considered.

Sometimes it is necessary or convenient to set up a large array of loudspeakers in such a way that the woofers and tweeters are placed at different distances from the listener. The effects of large (tens of milliseconds) interband arrival-time differences produced in this way, even on the intelligibility of speech, are very well-known and require correction by delay lines. However, the rest of our discussion is confined to "time-alignment" and "phase-coherence" as these terms are used in connection with arrival-time differences in the cases of monitors and domestic loudspeaker systems.

In the cases of even large monitors and domestic speakers, arrival-time differences cannot exceed a few hundred microseconds, and there is no experimental evidence in support of the notion that such a short delay, *by itself*, is audible.

Thus, if it were possible to separate the delay question from all other considerations, the answer to the question, understood in this way, would be: *NO*. However, moving the drivers axially also changes the phase-difference between acoustic out-

. . . continued on page 16 —

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— continued from page 12 . . .

puts and this, in turn, has a big effect on the amplitude of on-axis sum of acoustic output response, which is very clearly audible. Consequently, questions of the physical deployment of drivers cannot be separated from crossover network design.

This circle of ideas is very widely misunderstood by users and, if fact, by manufacturers. The reason for this is that the concept of the *acoustic center* of a driver is very abstract, and its accurate location requires elaborate instrumentation and computation. I believe that the second paragraph of my Preprint contains the first published exposition of these ideas, which is both elementary and correct, although those who have studied the technical papers of Richard Heyser in the *Journal of the Audio Engineering Society* will find little that is new.

Acoustic centers are physically aligned when that part of their on-axis acoustic phase-difference (using no crossover), *which is due to time-delay*, is reduced to zero by axial displacement. Lining up acoustic centers in a vertical line, in general, requires a stepped baffle. One can achieve essentially the same result by mounting all drivers on the same vertical baffle and introducing suitable time-delays. However, a genuine time-delay cannot be realized, even approximately, by any simple network.

The practical effect of lining up acoustic centers in a vertical line is to simplify the required network, at the expense of introducing new problems of diffraction control and box design. This may or may not be a favorable trade-off, depending on design objectives and the drivers to be used. The fact that a few loudspeakers with stepped baffles sound better than almost all other loudspeakers is not due to stepped baffles or "time-alignment." Rather, they sound better because their designers know what they are doing (and have made effective use of this technique).

In designing a network for drivers mounted on the same baffle, one must take into account the individual phase responses of the drivers, and the physical displacements of their acoustic centers. Since part of the total phase-difference which must be accounted for is, in fact, due to time-delay, it may be useful to the designer to conceive of part of the network as accounting for this delay. However, looking at the problem in this more sophisticated way does not guarantee superior results. Consequently, I, personally, have trouble attaching much importance to whether a designer tells me that he looks at the problem in this way and, therefore, designs "time-aligned" systems, or not.

On the other hand, I have not the slightest doubt concerning the fundamental importance of the circle of ideas we have been considering, in connection with Question 24, to the design of high-performance loudspeaker systems.

Let us return to the question of off-the-shelf crossovers. In sharp contrast to passive crossovers, active crossovers can be designed so that they can be continuously tuned in the field, in any number of parameters, to arbitrarily well approximate a true acoustic crossover. Further-

more, they can also incorporate variable delay lines to account for the locations of acoustic centers. For instance, my AES Preprint gives an example of such a flexible design, which can be built and tuned to yield a true fourth-order Linkwitz-Riley crossover, with only primitive instrumentation, given drivers for which the locations of their acoustic centers are known. All engineering design involves compromise. The more flexible we make the design, the more complex and costly it will be, and the more difficult it will be to set it up properly to achieve its full potential. My design, just mentioned, is not tuneable in crossover frequency and adding this feature —necessary to a general-purpose off-the-shelf crossover — would take it out of the realm of commercially feasible products.

A reasonable and attractive compromise, which is easy to set up, is a switchable 18/24 dB per octave electrical crossover, which can be tuned in crossover frequency, together with an equalizer. This makes available the three responses most likely to produce a fortuitous decent result (the 18 dB per octave output can be connected with either polarity). If, in addition, it is possible to physically adjust the horizontal displacement of acoustic centers, we have sufficient freedom to achieve a good result, even before equalization, with almost any pair of drivers whose responses overlap. Otherwise, we may require a short, adjustable delay line.

The answers to both parts of Question 21 is: *YES*.

The effects of polarity reversal are subtle, but clearly audible; see reference #5.

The audibility of phase question has been (hotly?) debated at length, for many years. At long last, in the most unexpected way, it has been recently answered by Fincham[7]. Fincham demonstrated, with a synthetic low-frequency tone burst signal which sounds something like a single drum beat, that the phase-distortion introduced by the conventional analog record/reproduce chain and a conventional loudspeaker is clearly audible to all listeners. The effect is much more subtle, but still audible, with suitable digitally recorded music signals. However, it cannot be demonstrated with standard recorded analog signals.

I believe that Fincham's discovery, together with the availability of digital signals which are free from low-frequency noise, has some interesting consequences for the short-term development of loudspeakers for the most critical applications. I believe that we can look forward to the disappearance of all sorts of vented boxes, transmission lines, etc., in favor of closed boxes which are electronically equalized to yield B2 (second-order Butterworth) responses with very low cut-off frequencies. Since, more generally, low-frequency distortion now appears to be more important than was formerly believed to be the case, we may also see more of motion feedback. In the absence of Fincham's discovery, of which I expect John Roberts was already aware when he wrote the Questionnaire, it would have made sense to ask Question 25: What box alignment do you prefer?

Finally, digital filter techniques free us from the amplitude/phase relations which we have so long believed to be cast in stone. Thus, in the slightly longer run, we can also look forward to digital crossover networks, based entirely on psycho-acoustic considerations, without regard for the engineering compromises we have had to live with for so long, and which will make everything I have just said part of ancient history.

I am neither a "Golden Ear," nor a "Meter Reader." The ear is the final arbiter of subjective quality. However, I also believe that the concordance between properly carried out precision measurements and subjective quality is very much better than is commonly believed to be the case, at least in some circles. There are, of course, some disparities, some of which are quite serious. However, it is my conviction that these are much more the consequences of the limitations of the present state of our knowledge of psycho-acoustics, than of anything else. We simply do not know what to look for in the measurements.

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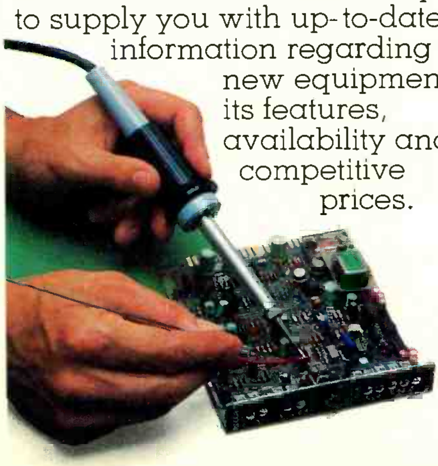

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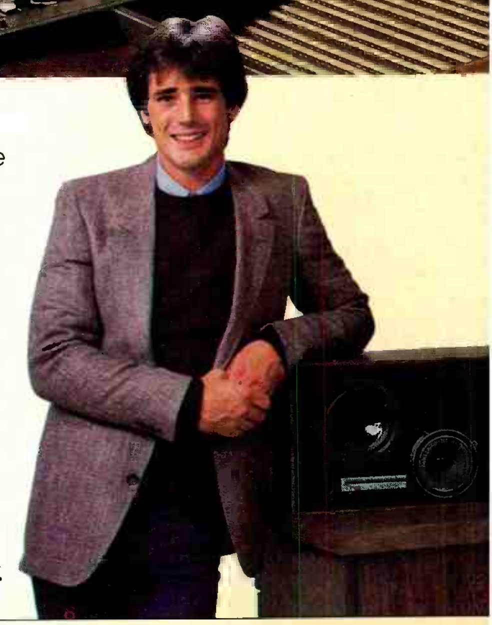

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Mythology/Zaustinsky — *continued . . .*

Aided Design of Loudspeaker Crossover Networks"; *J. Audio Eng. Soc.*; Vol. 30, pp 496-503 (July /August 1982).

4. "A target-function approach to the design of filters"; *KEFtopics*, Vol. 2, No. 1 (1978); and "Crossover Filters — An integral part of overall system engineering"; *KEFtopics*, Vol. 4, No. 2 (1980); KEF Electronics, Tovil, Maidenstone, England.
5. Lipshitz, Stanley P., Mark Pocock, and John Vanderkooy; "On the Audibility of Midrange Phase Distortion in Audio Systems"; *J. Audio Eng. Soc.*; Vol. 30 pp 580-595 (September 1982).
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7. Fincham, Laurie; "The Subjective Importance of Uniform Group Delay at Low Frequencies"; *AES Preprint 2056 (H-1)*, 74th Convention, New York, October 8-12, 1983. □□□

POSITIVE FEEDBACK

from: **Lance D. Abair**
Burlington, NC

Just a brief note to express my satisfaction with all of the great information *R-e/p* is feeding us. Between engineering and producing in a 24-track facility in Burlington, North Carolina, and performing in live concert situations, I find that you are covering both bases quite well, especially in the case of that new guy, Dave Scheirman. His articles always seem to include all of the stuff you always wanted to read, but no one seemed able to write.

Keep up the good work!

Editor's Note: Thanks for the positive feedback; it's always pleasing to hear from satisfied readers. By the way, it might be worth pointing out that our "new guy," David Scheirman, has been writing for *R-e/p* on a regular basis since the June 1982 issue, and also serves as our live-performance consulting editor. □□□

NOISE LEVEL MEASUREMENTS

from: **Don Davis, President**
Synergetic Audio Concepts
San Juan Capistrano, CA

With reference to the article, "Performing Meaningful Noise Level Measurements in Theory and Practice," published in the October 1983 issue of *R-e/p*, it is discouraging to find that Paul Buff, after six years, still doesn't know the difference between "signal level" and "voltage amplitude," and that he feels the use of the "dBm is irrelevant."

The technique he suggests is indeed irrelevant if correct "gains and losses" are to be obtained rather than "voltage amplification" figures. *R-e/p* and Paul Buff should correct this fundamental error so as not to mislead too many young readers.

Reply from: Paul Buff,
Technical Consultant,
Valley People, Inc.

What I believe we have here is an orchestration of the proverb of unknown origin which goes something like: "He who sits at the bottom of a well observing the sky is likely to become an expert in things over his head."

In reply to Mr. Davis' specific allegations, I will first say that, while indeed most contemporary audio engineers are not aware of the differentiation between the classic terms "signal level" and "voltage amplitude," or between "gains and losses" and "voltage amplification," it happens that I am well aware of the textbook differences. I consider that Mr. Davis has done a great disservice to the readers by simply making snide remarks toward myself and *R-e/p* about obscure points — points which the average reader will not understand, without offering appropriate clarification such that these readers might comprehend what he is talking about.

So, since Mr. Davis chooses the role of antagonist over that of educator, I shall do the job which he has left undone.

In the early days of electronics when most of the "classic" texts were written (i.e. the Twenties and Thirties) essentially all American systems were interconnected on the basis of "power matching." This simply means they were structured for the maximal transfer of power from one stage to the next. Now, the maximum transference of power occurs when the output impedance of the driving stage is equal to the input impedance of the driven stage. It was also common to employ isolation transformers between stages of a system, and transformers work best only at one specific impedance. *. . . continued —*

LETTERS

In the interest of standardization, and after many proposals and various standards, most of the industry (then primarily the broadcast industry) settled upon a standard where all outputs exhibited a 600-ohm internal impedance, while all inputs presented a 600-ohm load impedance. At the same time, the term "signal level" was used to define the amount of signal *power* present at any circuit point, while the terms "gains and losses" refered to the amount of *power* amplification or attenuation.

The term "dBm" was used to define "signal level" (signal power), with "0 dBm" representing a signal power level of 1 milliwatt, usually at the standardized 600-ohm impedance. Some factions of the industry chose to use other power reference levels, specifically 6, 10, 12.5 and 50 milliwatts. One common notation was simply "dB," which indicated relevance to 6 milliwatts in a 500-ohm circuit. Nevertheless, all of these terms were used to denote signal (or noise) power, *not* voltage.

When the specific specification of signal voltages, rather than power, was desired, classic texts employed the terms "voltage amplitude" and "voltage amplification," or "voltage amplification factor."

Now, onto the dB itself. From a textbook sense, the term "dB" can *only* be used to define the ratio of one *power* versus another, based on the formula dB=10log(P1/P2). Thus, from a classic sense, the expression of voltage amplification may not be made in terms of dB, since a 10-fold increase in voltage may, or may not, relate to a 20 dB (100 fold) increase in power, depending on whether or not the input and output impedances are equal.

So, then, according to the classic texts, and I suppose also according to Mr. Davis, the common op-amp circuit into which one puts 1 volt and gets out 10 volts cannot be stated as having a gain of 20 dB. Now, I know this is true, from a *purist* point of view. But, nevertheless, modern texts such as those issued by everyone ranging from the RCAs and Motorolas, down to the lowliest authors, commonly use the terms "gains and losses," and "signal level," as specifications of *voltages*, and express them in terms of dB, without reference to the power dynamics of the circuit or system.

Mr. Davis it would appear, stands as a lonely voice in the wind, obstinately refusing to acknowledge the chosen language of the contemporary masses.

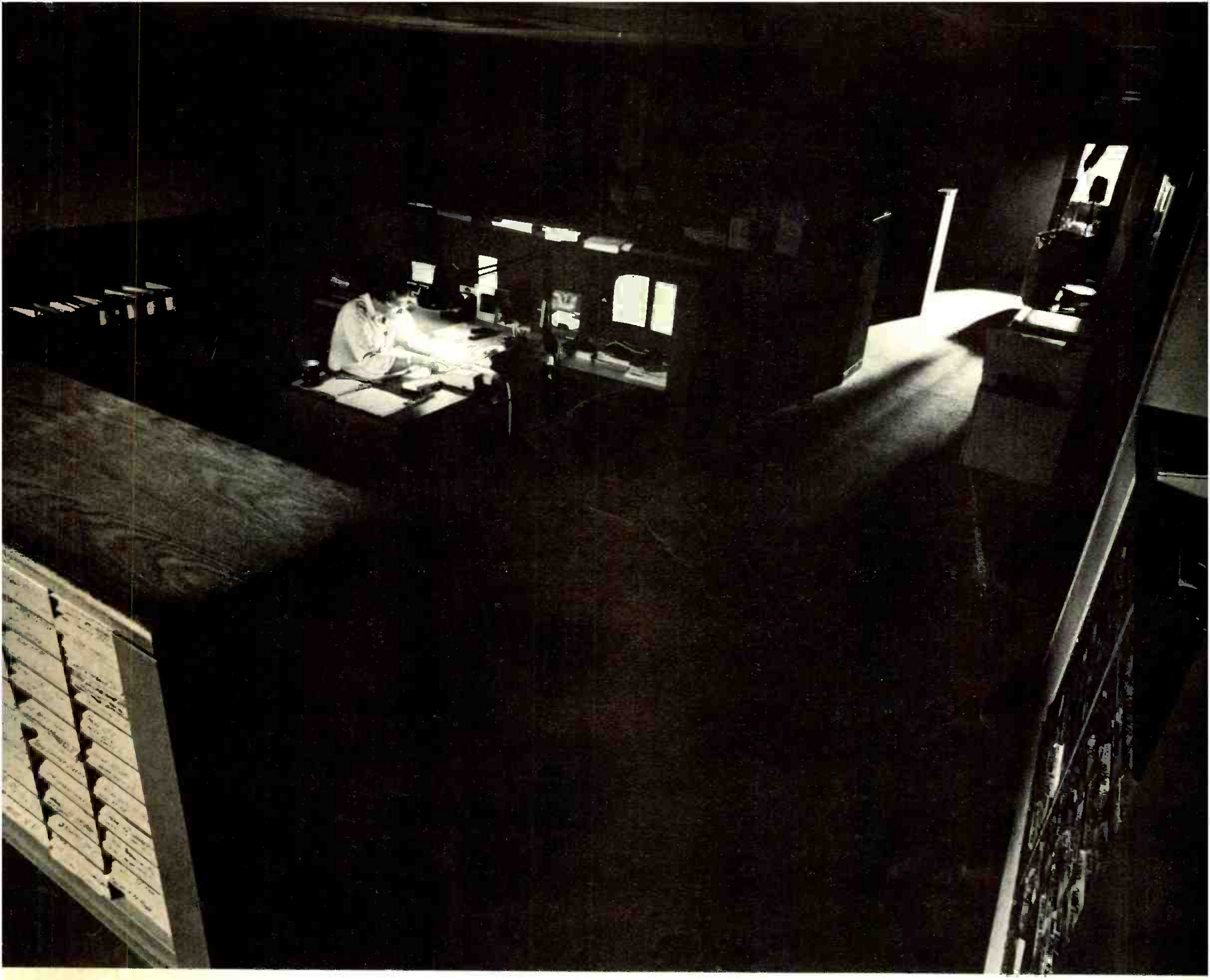
Finally, the term "dBm": I do *not* feel the term is irrelevant at all. It is fully relevant in the context of its definition — the measurement of *power* levels. In modern audio systems, however, power-matching has long ago given way to the optimization of voltage transfer. Near-zero output impedances typically feed high input impedances, and power transfer is rarely of consequence, save speaker amplifiers and old broadcast installations. When we measure signal levels (yes, "signal levels") we are usually, in fact, measuring voltages, and are almost never concerned with the actual power level. To wrongly express these voltage measurements in terms of dBm is not only incorrect and irrelevant, but is fully confusing to those knowledgable engineers who do comprehend the true meaning of the term "dBm."

I might point out that since the publication of my articles on this subject of some six years ago, there is a much greater awareness of the proper terminology of signal-voltage specifications, and that any number of manufacturers have chosen to amend their previous method of wrongly specifying voltages in dBm, in favor of the correct terminologies such as dBv re: 0.775 VRMS, dBV, dBU, and such.

Indeed, I believe the record will show that this writer, via the pages of ***R-e/p*** and other publications, has succeeded in making more clear a subject which Mr. Davis seems intent on muddying. So, let Mr. Davis teach his students the language of the ancients, but when they leave his

PRESSURE ZONE MICROPHONES® — A Correction

It would appear that gremlins struck during copy preparation for Stephen F. Temmer's "Letter to The Editor," published on page 13 of the December 1983 issue. Specifically, in the paragraph on page 14, in which the author considers the affects of direct and reverberant soundfields (referring to Figures 2A and 2B), the solid angles mentioned should have read "two-*pi*" and "four-*pi*" steradians, to designate a hemisphere and sphere, respectively. Our apologies for any confusion that may have resulted from this minor error — **Editor.**



LETTERS

tutorship let them begin to learn the "slang" adopted by the rest of the world, lest they remain in the well of archaism, along with their mentor.

Finally, in answer to the letter in the December issue from Steven Graham, who advocated using the European dBu and dBU specification, to prevent confusion between dBv (0 dB = 0.775 VRMS), and dBV (0 dB = 1.0 VRMS). I cannot agree more with what the writer says. The term "dBU" *automatically* defines what we are usually measuring today — a signal voltage relative to 0.775 VRMS. Unfortunately, if a manufacturer in America were to use this term today, without explaining its meaning, most spec readers would say "Huh . . . what does dBU mean?" Perhaps the best way would be for a manufacturer to state his specifications with "dBU*," then explain in a footnote that dBU and dBv re: 0.775 VRMS have the same meaning. □□□

EXPOSING AUDIO MYTHOLOGY

Laying to Rest some of the Pro-Audio Industry's more obvious "Old Wives Tales"

by John Roberts

This month we'll review the results we have received so far from the opinion survey and questionnaire that was published in the October issue. Part II will include a look at the non-ideal characteristics of capacitors.

The Survey

Before we get into the numbers, I'd like to offer a word of caution: the results I've received from the October questionnaire do not represent a scientific sample, and the questions were not properly framed for easy yes/no answers. Therefore, the following statistics may not be representative of any larger group. Nevertheless, the results will perfectly describe the answers of the industry leaders that did respond.

Question #1: 25% of the respondents favor the sound of digital audio; 50% do not; and another 25% didn't vote. While in previous columns I have already covered digital audio in general terms, I am more than willing to re-open the discussion. Perhaps a few of the 50% who do not favor digital audio can tell us what they didn't like about the sound; specific examples of digital weaknesses would be useful.

#2: The poor transformer did worse than digital; 63% voted No, with nobody favoring the sound. (This was a trick question, however, since a good transformer circuit shouldn't have a characteristic sound.)

#3: 38% voted for gold-plated jacks, while 50% voted nay.

#4: 88% voted that speaker wire does make an audible difference, and 12% said No. (I wonder if they read my column in the same issue?)

#5: 63% said slew rate does make a difference, and 12% answered correctly that it doesn't.

#6: 50% answered that negative feedback improves sound, and 50% said it doesn't.

#7: Nobody voted against capacitors being audible, and only 12% didn't vote. (See Part II of this issue's column for my opinion on the matter.)

#8: The spread was 25% for 6 dB per octave, 25% for 24 dB, 12% for 12 dB, and 12% for 18 dB per octave crossovers. [See this month's Letter's column for some additional input on the subject — *Ed.*]

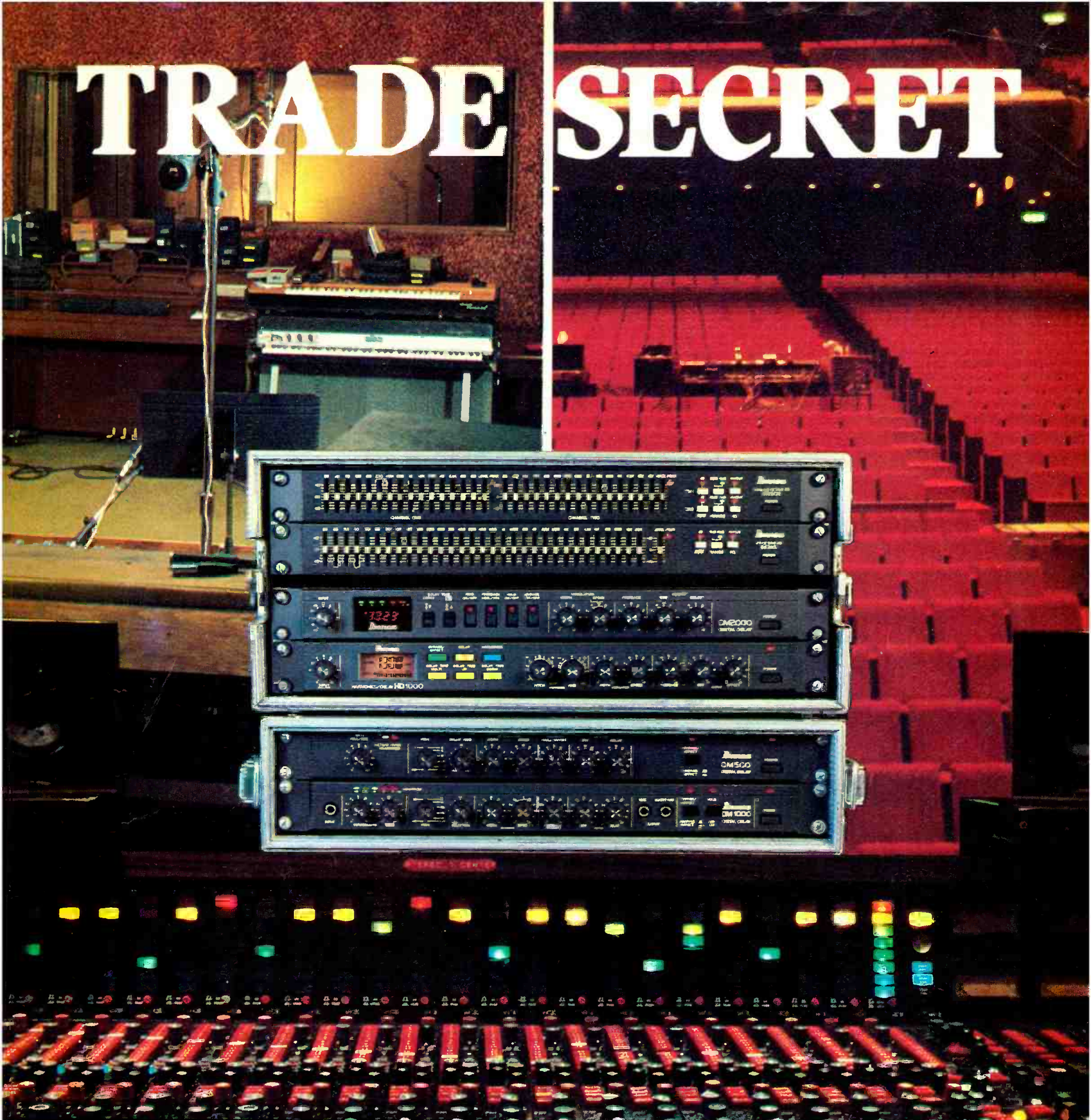
#9: 75% voted against tubes, and 25% voted for. (I wonder if it was the same 25% who voted for digital?)

#10: A perfect 50/50 split for discrete components. (I agree with half of you.)

#11: 75% favor moving-coil cartridges, with no votes against. (I'm not going to vote against.)

#12: 75% feel half-speed mastering makes a difference, while 25% don't.

#13: We actually got some votes for adding weights. (Could somebody out



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there tell me what that's supposed to do?)

#14: 88% feel that low-level signals can be affected by wire type, and 12% don't. (I know where I'm not recording.)

#15: While 50% voted that an unused speaker can cause audible degradation, and 38% said No, I still don't recall ever seeing a control room with less than two pairs of monitor speakers; many have more.

#16: 38% favor MOSFET power amps, and 25% favor bipolar. (I wonder what the other ones like?)

#17: This question got the weakest response, with barely half the people voting. Even then it was a tie with 25% favoring +4 dB and 25% favoring -10 dB systems.

#18: Another strong turnout, with 25% for and 38% against dubbing tapes backwards.

#19: 75% (of everyone who answered this one) voted that the type of jack or interconnects can be audible.

#20: Surprisingly (to me) the response was 100% Yes that power-supply regulation is audible.

#21: 63% voted Yes to 25% No, regarding the audibility of phase shift. (See the April 1983 column for my opinion on the subject.)

#22: 25% of the respondents voted that proximity of speaker wire to structural surfaces can be audible, 50% voted that they aren't, and 25% didn't vote. (I'd also appreciate it if somebody would clue me in on this one.)

#23: A strong 88% voted that speakers can image wider than their placement, with no objections.

#24: 62% voted that time alignment of speakers is audible, to 12% voting No.

On the subject of "Golden Ear" versus "Meter Reader," it was an almost even split; 55% characterized themselves as Meter Readers, with only one voting "other."

CAPACITORS: IS THERE A DIFFERENCE?

Capacitors rank right up there with speaker wire for possessing near mystical (or should that be mythical?) power over a system's sound quality. As with speaker wire, a handful of companies have sprung up to market "improved" high-tech capacitors to the poor souls saddled with sonically deficient components.

So far, every subject we've studied in this series has had a basis — however meager — in fact. This topic is no exception. Like the others though, there is also much spurious opinion. My favorite capacitor story is about an adventurous audiophile who replaces all the capacitors in his 25- to 30-year-old tube pre-amp with new "......"-caps. (I couldn't think of a funny name that wasn't already being used; readers are more than welcome to supply their own.) This "born-again" pre-amp sounds dramatically better, the user claims, living proof that oxygen-free, gold-plated, Teflon-coated capacitors sound better than your, off-the-shelf garden variety. Or is it? I have had electrolytic capacitors fail (open-circuit) from loss of electrolyte at the tender age of 17 years, so I would *not* be surprised if 30-year-old ones show enough deterioration to be out of spec, and audibly significant.

I suggest that had our friend merely replace has capacitors with fresh equivalents, he would have still heard a great improvement. Assuming the pre-amp was properly designed in the first place, slight further improvement could be gained from substituting newer-type caps. And these don't have to be expensive. For small amounts of capacitance, I'll put a 15-cent polystyrene up against any type at any price!

Non-Ideal Characteristics

A casual inspection of any piece of electronic gear will reveal several different types of capacitors. Capacitors have many modes of non-ideal behavior, and different types optimize one or more in favor of the others. Besides the expected capacitance (Zc) real capacitors also exhibit:

Zr or ESR: Equivalent Series Resistance. This is resistance caused by leads and internal construction, and is modelled as a resistor in series with an ideal capacitor.

Zl or ESL: Series Inductance. This looks like an inductance in series with the ideal capacitance.

Leakage: This looks like very large resistance in parallel with ideal capacitance.

DA or Soakage: Dielectric Absorption. Because of the way in which charge is stored and distributed about within the capacitor, a "memory"-type effect occurs. After release of short charge or discharge cycles the capacitor returns partway to its initial state of charge. DA is modelled as several different small RCs in parallel with the ideal capacitance.

DF: Dissipation Factor. DF is a function of the reactive and resistive components in regulators.) It is also common to locate ceramic discs with values between 0.01 and 0.1 mFd near IC-power pins, again for low impedance at high frequencies.

Equalizers and Active Filters: Film capacitors such as polystyrene and polypropylene are often the first choice for such designs; their near-ideal performance will deliver near-ideal sound quality.

Because film capacitors become unwieldy for larger values, it is sometimes necessary to use electrolytics for DC blocking applications. A successful technique to minimize expression of the electrolytic's non-ideal characteristics is to tune for a highpass pole several octave below the lowest frequency of interest. Voltage coefficient and DA cannot do any damage if there is no AC voltage generated across the capacitor. Likewise, a one-ohm change in impedance versus frequency will be down 100 dB to a 100 kohm load.

Passive Crossovers: Now we have to be careful. That same one-ohm change in series impedance will be significant to an eight-ohm speaker. Since there will also be audio developed across the capacitor, DA and voltage coefficients will be expressed. In fact, every non-ideal characteristic with the possible exception of leakage can be a problem.

Film capacitors large enough to handle tweeter crossovers can be found, but low to mid frequencies will have to use those dreaded electrolytics. It is worth noting that not only do different electrolytic dielectrics behave differently, but similar dielectrics of different manufacture can be vastly different. Generally, tantalums will have lower leakage, and lower ESR but higher DA. The trend towards higher frequency switching DC-to-DC convertors has encouraged manufacturers to reduce ESR in aluminums.

Similar to the power-supply application, to improve their high-frequency performance it is possible to paralell electrolytics with smaller film types (no ceramics please). However, there is no design trick (that I know of) to minimize expression of DA and Voltage Coefficient; one must find a capacitor manufacturer that optimizes for such applications. Most electrolytics are optimized for low cost, low leakage, or small size. None of which are very important to the crossover designer.

To answer the question: Can a capacitor be audible? Yes, if you are using the wrong type for the job, and Yes maybe, if the capacitor is older than you are. ■■■

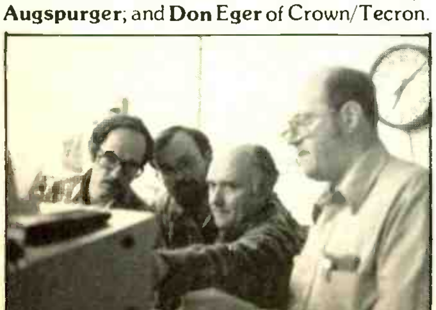
news

NEW IN-CAR STUDIO QUALITY MONITORING SYSTEM TO FEATURE EXR, JBL AND CROWN EQUIPMENT: ACOUSTIC DESIGN BY GEORGE AUGSPURGER

The Mobile Studio™, a new option for the Chrysler Plymouth Voyager van, is described as the first studio-quality monitor sound system to be offered in a production vehicle. Developed jointly by Conceptual Engineering Corporation, which was responsible for the system's overall design, and Cars & Concepts, the company that developed suitable loudspeaker enclosures to George Augspurger's designs, and also handles sound treatment of the vehicle's interior, the Mobile Studio features newly developed Crown power amps, JBL loudspeaker units, Proton radio and cassette deck, plus EXR effects units — the first time that EXR's Projector psychoacoustic processors have been made available to the consumer market.

The specially designed Crown amplifier package comprises six 50W, 12 VDC units that is heat sunk to the van's frame. All loudspeaker enclosures, which are configured around JBL bass, mid-range, and HF drivers, plus acoustic modifications to the basic vehicle interior, were built to George Augspurger's design specifications.

The Mobile Studio design team utilizing a Tecron TEF analysis system to check out acoustics of vehicle interior: (L-to-R) **John Wilson** of Conceptual Engineering; **Paul Gilson** of EXR; acoustic consultant **George Augspurger**; and **Don Eger** of Crown/Tecron.



As well as providing a high quality listening environment for producers, engineers, and artists travelling on the road, the vehicle can also be configured, with the addition of supplemental equipment, to serve as a mobile control room for live recording and mixing. By way of an example of the type of recording and mixdown hardware that can be accommodated in the van's acoustically treated interior, EXR currently has under construction a Voyager that will include a Neve 5432 eight-input stereo console, dbx Model 700 CPDM digital audio processor and companion RCA VHS videocassette deck, Lexicon 224X digital reverb, EXR EX4 and SP2 Projector audio processors, and a MIXMIX Master-Room reverb. Video equipment will include a JVC ¾-inch U-Matic VCR, and Proton monitors.

NEWS continues on page 157 —

THE CREATIVE INTERFACE

COMPUTERS AND MUSIC PRODUCTION

by Shelton Leigh Palmer

Everyone has seen album credits that read: "Synthesizer Programming by John Doe," or "Drum Machine Programming by John Q. Public." Most people have seen every kind of credit from "Case Slapper" to "High Grunts." But there's a new kind of production credit coming: "Creative Computer Programming by . . . "

By now, I'm sure *R-e/p* readers are familiar with a number of computer-controlled synthesizers and computer-based musical instruments. The technologies of Sequential Circuits, Oberheim, New England Digital, Fairlight, Roland, Yamaha and many more have become commonplace in the modern recording studio. [See Paul Lehrman's article, "Computer Controlled Synthesizers," published in the December '83 issue — *Ed.*] But most people forget that all of these instruments are controlled by computers, and therefore require a computer programmer to make them work.

Computer programming in music production occurs at many different levels. At the "lowest level," machine language, the programmer creates a set of instructions that tells the microprocessor how to run the system in which it lives. This includes: tuning functions, patch storage, patch editing functions, keyboard multiplexing, etc. (When I say "lowest level," I am referring to levels of programming with regard to music production, not complexity of design or use. If you've ever written a program in machine code, you know that there is nothing low-level about it.)

At higher levels, computer production programmers concentrate on inter-instrument communication, (sometimes called interfacing), and computer-controlled synchronization techniques. This is the "computer programming" that album credits are made of; interesting and unique methods of patching all of one's synthesizers and sequencers together to make one monster musical instrument.

Any synthesist will tell you horror stories about the problems of getting various machines to "talk" to one another; the problem is as old as synthesizers themselves. Although we have had computer-controlled synchronization for many years, this writer has seen many synthesists use up six tracks of their 24-track tape for sync pulses and click tracks. This has always been unacceptable to me as a composer/producer, so I looked to the computer for alternative methods of patching and synchronization.

I remember my first efforts toward electronic synchronization about eight years ago. The system included a few monostable multivibrators, a master clock, a 6500 Series microprocessor, and an incredibly large (at that time) 4K of RAM. All of this was thrown together on a hand-built printed wiring card to form a basic, dedicated single-board microcomputer. The system was programmed in hex code and it worked very well for what it was: I was able to take my ARP 2600, Odyssey, Omni, Moog System II, Minimoog and Univox Rhythm Box, link them together, and make them "talk" to each other. The sequencer/editor program was very primitive due to the size of available RAM. So the search for a "better way" began.

Within a few years of that first glorious invention, we found a solid 16-bit minicomputer, some very serious DACs (digital-to-analog convertors) and ADCs, and we had most of our synchronization problem beaten. The music I composed and produced in those years sounds very similar to the tracks that are becoming popular today — sort of "electronic" with interesting complex rhythms. As you may have guessed, this technology is the mother of "Electro-pop."

As with the theory of evolution, similar innovation was occurring all over the world. Sequential Circuits, Oberheim, and Moog were busy inventing new analog/-digital hybrids. At the same time the home multitrack recording studio became a reality. Suddenly, musicians had a choice: Learn the technology or be at the mercy of those who did.

The smart synthesist/composer added engineering programming and producing to his list of required crafts. The idea that people were working across disciplines slowly gave way to the idea that there was

Composer/producer, **Shelly Palmer** is president of Shelton Leigh Palmer & Co., Inc., a New York-based music production company specializing in computer-controlled electronic music. His credits include hundreds of television and radio commercials, as well as music for a number of feature films. An alumni of NYU School of the Arts, he returned there to teach Audio for Television: Applications and Techniques. Along with his commercial business, Palmer is currently producing a new album, rock video, and interactive video game.

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only *one* discipline: Electronic Music. Some people will argue that such a label refers to a specific musical style — nothing could be further from the truth. Any music that will be reproduced by electronic means is "electronic music." Whether you are an acoustic purist or a punk rocker, if you finish on tape, the music is "electronic." Each note to be played becomes important only as it relates to the final mix, not as it is originally heard over the studio monitors. Each note must be produced within the context of the eventual listening environment — this is the gospel. The greatest bass line ever written means little if it sits below the sonic range of the playback system.

The Computer in Production

Of the many uses for computers in music production the two most common are Patch Storage, and Sequencing. Patch storage is pretty well covered by the commercial equipment manufacturers. Since all of the instruments are different in design, the patch data cannot be standardized. Although the data formats are different for each device, MIDI (Musical Instrument Digital Interface) will transmit Patch Address Status from one instrument to another in sequential form. So, if you have patch Right 1-1 on your Prophet T-8, it will call up Patch 01 on the Yamaha DX-7, etc.

There are a number of contenders in the race to create the "ultimate" sequencer/editor package. The first hardware/software package that comes to mind is the Script Music Composition Language by New England Digital, written for the Synclavier II; it does almost everything. If NED's offering is out of your price range, there are packages available for the IBM PC and Apple II+. Either of these microcomputers with some additional memory, a hardware interface (eg: UART, timers, etc.) and some software can get you started for the cost of the Synclavier keyboard unit alone. You won't get some of the higher function computer music features, but you may not need them.

Typical Production Set-up

Let's look at the role of computer in the production of an average post-score. In this example we'll use my standard set-up: Synclavier II, Sequential Circuits Prophet T-8, Yamaha DX-7, LinnDrum, a highly modified ARP 2600, a Lyricon interface, and a couple of Apple II+ microcomputers.

Before production can start, I must compose the track. I don't use a computer to compose the music (we haven't built M-5). However, I do use a program called The Master Click Program to help me with the film math. After I hear the track in my head, I choose a tempo range for the spot (for example: 120 to 130 beats per minute). In this example let's say there are 10 hit points in the spot. (A hit point is a specific frame of the visual that must be hit with a musical accent.) I type the hit-point data into the Apple computer. The Master Click Program tells me everything I need to know about the timing of the music: it selects a best tempo based on my own preprogrammed musical preferences, and prints out a report showing all of the important aspects of each hit point (eg: location, description, duration, etc.). Master Click and an Apple are an invaluable combination for post scoring; the time saved is incredible.

Now that the music for the spot is written, I must deal with the sound effects. Some of these may be constructed on the Synclavier, T-8, or DX-7, while others may be sampled or sampled and re-synthesized on the Synclavier. However they are produced, the effects eventually must end up on tape at a specific frame of the visual.

The Script Music Composition Language from NED is part of the software that comes with the Synclavier II. Like most commercially available hardware and software combinations it is not out-world compatible. We have modified the hardware and software to meet our needs. With our proprietary Music Composition Language we can type note or SFX locations into the Synclavier terminal as follows:

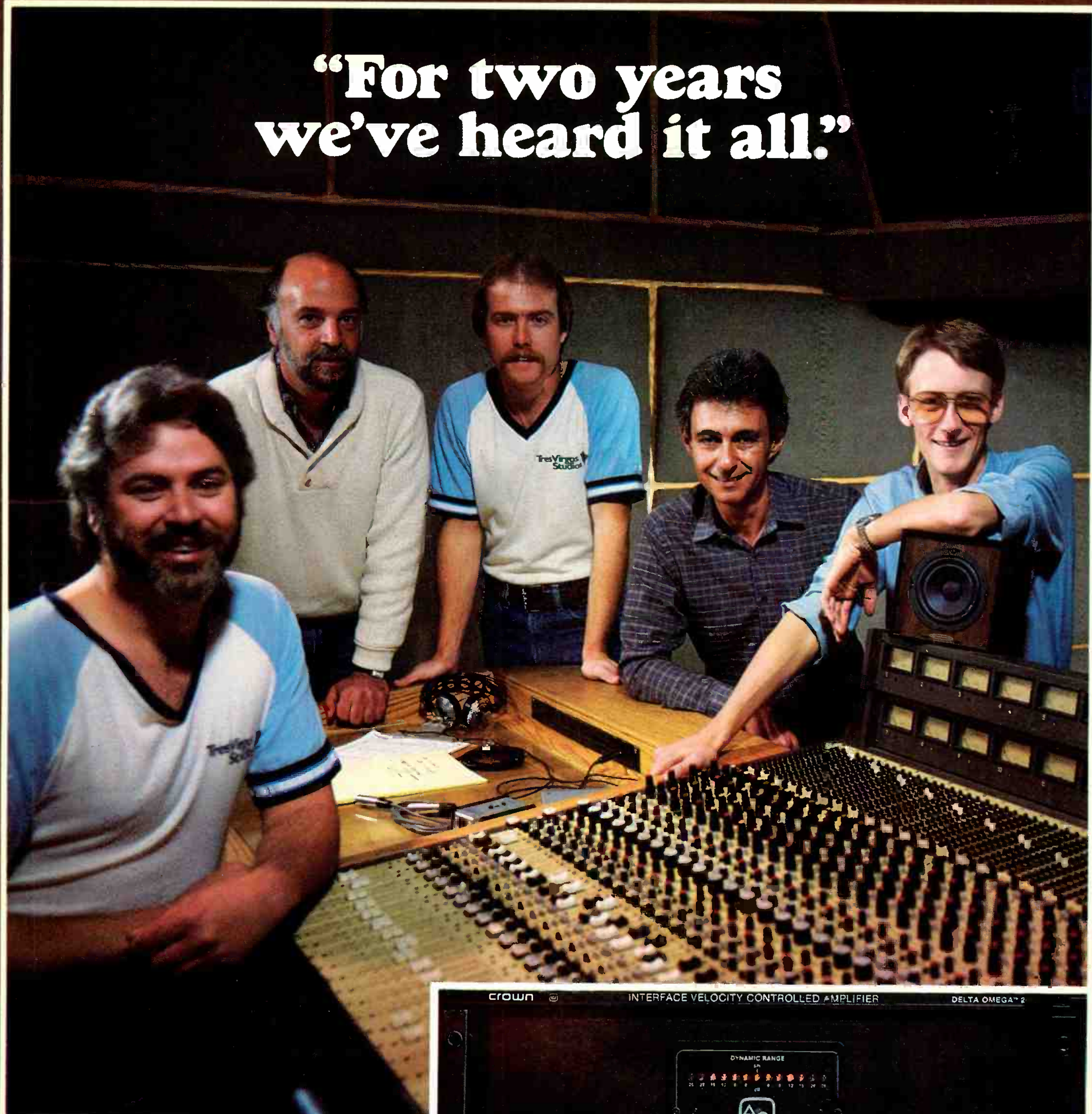
Notelist Using 3-4/S1

00:01:03	*C3*	*00:00:20*	*100.0*	*73.00*	*86.00*
(a)	(b)	(c)	(d)	(e)	(f)

a: Location of hit point in SMPTE Timecode
b: Pitch and Keyboard Location
c: Duration of note in SMPTE Timecode
d: Volume Multiplier
e: Articulation Multiplier
f: Timbre Multiplier

You can set a default value for Maxi-

. . . continued on page 152 —



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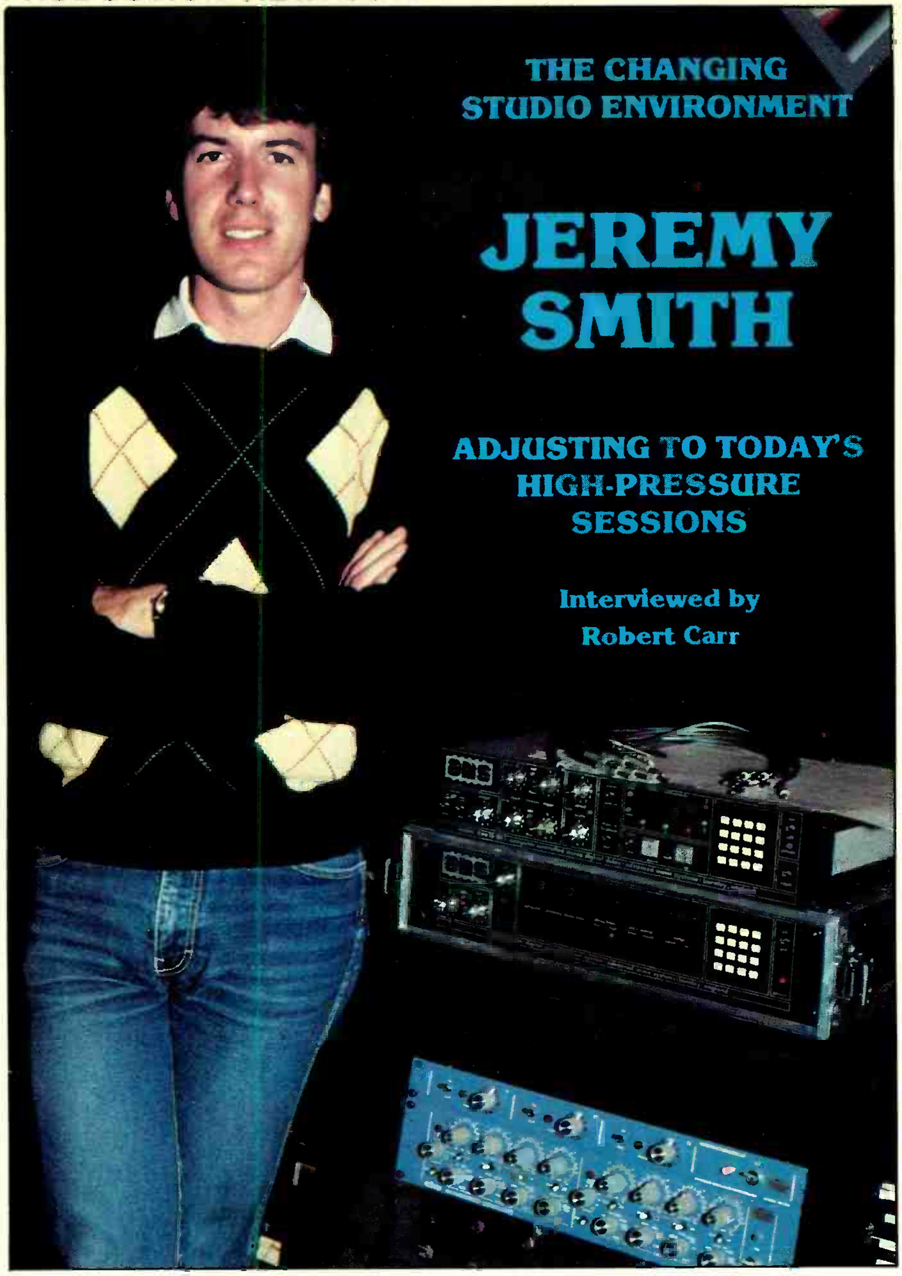


THE CHANGING STUDIO ENVIRONMENT

JEREMY SMITH

ADJUSTING TO TODAY'S HIGH-PRESSURE SESSIONS

Interviewed by Robert Carr

Photography by Kathy Cotter at Motown Studios/Hollywood

Jeremy Smith broke into the recording business when Trident Studios in London gave him his first job — making tea and sandwiches for his co-workers, and then "I graduated to making coffee," he laughs. The pay wasn't very good for 10-hour days and seven-day weeks, but the experience was irreplaceable, given the stature of his mentors: Ken Scott, Robin Cable, Roy Baker and David Hentschel. After a six-year apprenticeship at Trident, Smith came to the U.S. in December 1978, looking for opportunity in California. Since then he has worked as an independent engineer with an impressive list of artists, including George Benson, Melissa Manchester, Leo Sayer, Aretha Franklin, Neil Diamond, and many, many more. Smith credits much of his current success to his association with producer, arranger, and Atlantic Records VP Arif Mardin, with whom he works regularly. Also on his agenda are plans to open a studio in Los Angeles with record producer Dennis Lambert.

The conversation that leads into the interview started out with reminiscences about Smith's earlier recording experiences in England, when session budgets were a little bit larger and studio costs weren't as high as they are today. The less-pressurized atmosphere nurtured a healthy attitude of experimentation, which he feels is slowly, but surely, disappearing from the current music scene in America.

R-e/p (Robert Carr): How would you characterize the various changes in studio sessions that have happened over, say, the last five years?

Jeremy Smith: A lot of dates that I do now feature a solo artist accompanied by session musicians, so I have to work within the parameters that the players, who are pretty much hired for their sound, and the situation dictates. I have to get the whole thing rolling in a half hour or an hour at the most. I *don't* have a day to spend on a drum sound.

R-e/p (Robert Carr): Which sounds a lot like a jingle date.

Jeremy Smith: Yes, like jingles, the approach has a lot to do with the costs involved in making a record. The studio may be $130 an hour or more; the engineer's fee is added on top of that; and most musicians get double-scale. Expenses add up *quickly*, so you had better get on with the session, and get it happening. Otherwise the producer and artist will want to work with someone who *can* get it happening a little faster.

In the earlier part of my career, while still in London, we had the luxury of spending more time getting the basic sounds, because the clock wasn't continually beating us over the head. In some respects, we were a little more aware of getting sounds that excited you.

R-e/p (Robert Carr): Do you think that the current budget restrictions are taking a toll on the quality of the music?

Jeremy Smith: In some respects, yes. There are definitely things that happen on a date that, given more time, I would change. But the overall vibe, soundwise, I try to get on the tracking date. I hate pushing up a fader while I'm mixing, and thinking, "If only I'd spent another 10 minutes, or another half hour . . ." I usually stick out for a little more time if it's really necessary, and if I think I can get away with it. But I *don't* like wasting time. If something sounds good to me instantly, why spend the time fussing with it?

R-e/p: How do you decide on the particular sounds that are needed for an album?

JS: The producer pretty much leaves it up to me, and I work with the musician; *he* is really the source of the sound. Take Phil Collins' drum sound, for example —all an engineer has to do is embellish it. Somebody like [session keyboardist and arranger] Robbie Buchanan, who has had several modifications done to his piano, gives you a sound that needs nothing else done to it.

R-e/p: Isn't it almost essential to have standard miking techniques, or choice of mikes that you can depend on as starting points?

JS: Sure. When I record a bass drum, I use an EV RE-20 or a Sennheiser 421; one or both of these *always* work for me. An RE-20 has a better bass response,

For additional information circle #20

SESSION RECOLLECTIONS

Melissa Manchester's *Emergency:* Jeremy Smith recorded all the basic tracks and mixed the song "City Nights," on which Robbie Buchanan programmed a LinnDrum Machine, operated a Vocoder, and played Roland Jupiter 8, Minimoog bass, and Fairlight CMI synthesizers. The distinctive sound on the snare was obtained, Smith recalls, by running the Linn through a Lexicon 224X digital reverb set on a short room echo.

"I think Melissa's voice sounded the best on the tune I mixed ["City Nights"], primarily because it was done on a Neve 8108 console, which is far more open sounding. The rest of the tracks were mixed on [another] console, which is not as transparent.

"The voice was mixed back in relation to the music. If you listen to a hard-driving group like Genesis, it's often difficult to flush out exactly what they're singing about. In that kind of a situation, the voice is used more like another instrument, rather than a crack lead vocal. The tracks on Melissa's album, such as 'Johnnie and Mary,' are quite powerful, so you can't have the guitar or drums way back, and the voice way up front; the track will sound very *weak*. Because the music track has to be burning, like on a live performance, the vocal *has* to be mixed at a level that's similar to the high energy of the track, so the two are proportionately correct.

"On 'Johnny and Mary,' drummer Carlos Vega played a Simmons [electronic] snare drum and toms, all of which were taken direct. The high-hat, bass drum, and cymbals were traditional acoustic instruments. Carlos put a lot of white noise in the Simmons' snare sound. We also put a lot of room sound on the bass drum, which was recorded at Music Grinder. When [producer] Arif [Mardin] mixed it, he probably put a short [EMT] 250 or 251, or short Lexicon delay on it."

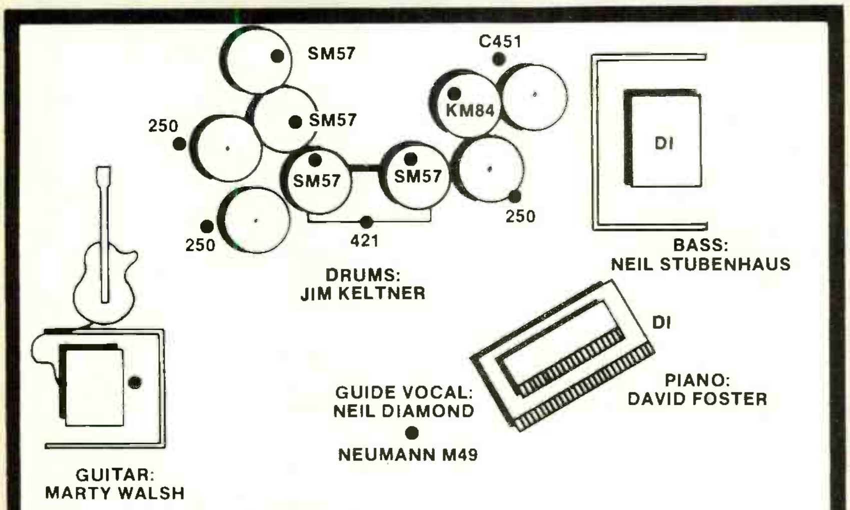
George Benson's *In Your Eyes:* Smith recorded and mixed most of the tracks on the album, except "Lady Love Me" and "Inside Love."

" 'Lady Love Me' was recorded at Music Grinder, with the majority of the rest of the album recorded at Atlantic Studios in New York. It's really a shame, [because] I would have loved the opportunity to do one of the best cuts on the album. I was asked to do it, but couldn't, because I was working on something else.

"We cut the track 'Feel Like Making Love' initially with real bass. Robbie [Buchanan] overdubbed Moog bass, and then Will Lee overdubbed some fills. But the majority is all Moog bass. Steve Feronne [drummer with Average White Band] played an acoustic drum kit.

Neil Diamond's *Heartlight:* The basic track is David Foster on piano, Neil Stubenhaus on bass, Jim Keltner on drums, Marty Walsh on guitar, Craig Hundley on synthesizers, and Paulinho da Costa on percussion.

"We overdubbed acoustic guitars, synthesizers, percussion, strings, and punched in one vocal line. David Foster played Robbie Buchanan's piano, which was taken direct. It has one straight side [left], and the other side [right] had some kind of chorus across it. EQ was flat. It's all the



ROOM LAYOUT FOR NEIL DIAMOND'S "HEART LIGHT"
Engineer: Jeremy Smith
Studio: Ocean Way, Hollywood

— *continued from page 37 . . .*

JEREMY SMITH

echo; I'd prefer a great drum sound, and *then* use huge amounts of echo to make it bigger and, hopefully, better [laughter]. I don't believe you can really "manufacture" sounds. The sound has to come from the room, and the engineer has to know how to transfer the sound from the room to the tape machine. Trying to do it purely with EQ gives you an electronic sound.

R-e/p: I assume that you keep the EQ on the tracks pretty flat?

JS: I try to, but I don't mind using EQ either — that's what it's there for. For my taste, I find that drums need the most EQ. As much as the snare might be cracking, and [sound] bright and big in the room, it doesn't transmit that way up a microphone, regardless of which [mike] you use. *Nobody* listens to the bass drum with their head stuck in the middle of it, and that's basically what the mike is doing; the microphone is *so* close that the sound doesn't get a chance to grow. Therefore, you have to help it along with equalization.

R-e/p: I've noticed while sitting in on sessions that English engineers tend to use more equalization than American-trained engineers. Would you agree with that assessment?

JS: I think so. For a long while, the English approach has been towards getting sounds that are a little brighter, more open, more ambient, more orchestral. The American tradition has been the use of a lot of close miking for a very "Up-in-Your-Face" sound. For my taste, that's a little colorless . . . flat sounding. The issue is really *so* subjective. There have been some great records made in America and in Europe. But, overall, I think the English have produced some "ballsier" sounding albums.

In all honesty, I haven't seen many engineers here in the States use 12 dB of equalization on a bass drum. I'll do that without thinking about it. If I want a bass drum to blow you head off, I don't feel at all hesitant or embarrassed about using lots of EQ. It's there to be used.

Basically, I think that English-made consoles are more transparent than American consoles. That openness allows you to use more EQ with shelving than you would on one built here.

R-e/p: Which brand of consoles do you prefer to work on?

JS: I like the Trident TSM, and the [older] Series A, for example. The Neve 8108 I like a lot. In fact, the Neve NECAM system is the best computer on the market. I hate VCAs, although somebody [from a leading U.S. console manufacturer] told me that, technically, it's a better signal path than a passive fader. Well, "technically" doesn't matter; what matters is the *sound*. I can hear a VCA — there's a lack of transient

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did some overdubs. The producer, Arif Mardin, who is also a vice-president of Atlantic Records, lives in New York, and comes out here to Los Angeles to record. Because he's sometimes working with more than one artist at a time, he doesn't always know exactly when he'll be mixing a specific project.

After I recorded the basics on George's album, I had to go off and record something else, so Gary did some vocals, a few keyboard overdubs, and recorded all of "Lady Love Me," except the strings which I handled. The same situation happened with Michael O'Reilly; apart from one cut ["City Nights"], which I mixed, everything else was mixed in New York at Atlantic.

Although Arif and I generally get the tracks completed before he goes back to New York, there may be little lines that he needs to add, which he does with musicians in New York. Michael records all those overdubs. Obviously, I'd like to do the whole project from day-one because that maintains a certain continuity. But, quite often, scheduling and financial consideration doesn't always make that possible.

If I'm already doing a project for someone who is not a major artist, and a major artist or producer asks me to do some work for them at the same time, chances are that I will have to decline. Because of my own personal code of ethics, I can't get a sub to do what I'm doing on the first project, and then run off to do the hit. Occasionally, I've had to turn down some very nice things. It's very flattering, but I can't do everything. I figure that the person I have to turn down will call back, because they understand the importance of someone else's project, and they know that I wouldn't bail out on them if the situation was reversed. They trust me more. Conversely, the large majority of the people I work with, especially Arif, are very straight forward and never let me down. I prefer to work with four or five people that I really respect and enjoy, rather than do dates with everybody in town, and enjoy only 50% of it.

R-e/p: I would think that kind of attitude translates into the quality of the job you do. Don't the tracks reflect that loyalty?

JP: I think it comes across in the music. Even if an album doesn't do amazingly well saleswise, I still liketo feel that I've done a good job. If the feeling in the studio isn't good, I don't feel good, and then I'd rather be home painting my house.

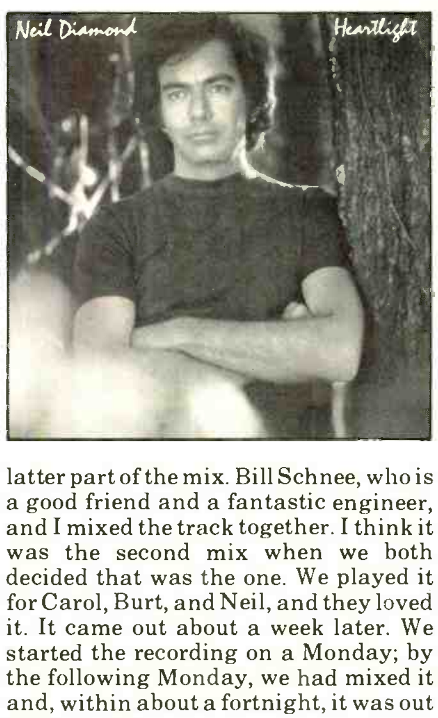
R-e/p: Getting back to the original train of thought . . . there were several producers on Neil Diamond's Heartlight *album, as well as a half-dozen engineers. Why so many producers?*

JS: In that case, Neil had to get a lot done in a short space of time, so he had several studios, producers, and engineers on the go at one time. I worked on the single ["Heartlight"] with Burt Bacharach and Carol Bayer Sager, who also co-wrote the song. Then Michael Masser, Alan Lindgren, and Richard Bennett produced the cuts they wrote.

In general, I think artists are searching these days. They feel that one producer may be best for one cut, and another producer would be just right for the next song. Maybe they think that if they pool all those talents, they'll automatically have a hit album!

R-e/p: What is it like working with three co-producers — Burt Bacharach, Carol Bayer Sager, and Neil Diamond — on the single "Heartlight." Did Neil have a lot of say in how the mix came together?

JS: Neil was in the studio for only the



latter part of the mix. Bill Schnee, who is a good friend and a fantastic engineer, and I mixed the track together. I think it was the second mix when we both decided that was the one. We played it for Carol, Burt, and Neil, and they loved it. It came out about a week later. We started the recording on a Monday; by the following Monday, we had mixed it and, within about a fortnight, it was out in the shops.

The vocal on "Heartlight" was a guide vocal that Neil sang as the track was going down. We repaired just one small part, and that's all. Burt's experience tells him when the feel is right, or when something is a little out of tune. Why mess with something that's so good?

Generally, in the recording process, the artists and producers give me the freedom to do what I want to do. If someone were to sit next to me and tell me they want this here, and this to sound exactly like this, sure I could do that. But I think it's a waste of time to employ somebody like me. I would rather have Burt or Carol or Neil sit down with me and say, "In the second verse we need this a little bit louder. And then in the first chorus we need this a little louder, and a little more echo on this." I'm totally open to direction. As much as I care about the process from A to Z, the bottom line is that the record is *not* mine.

But what I do is a very important additive to the artistic process. Engineering has come a long way from just twisting knobs and punching faders. I have a certain amount of input with my regular clients; we work as a team, so I feel very much a part of a project. I get inside the music to make it work . . . to make it sound good. To me, sound is very much like a painting with lots of diffferent colors; it's like seeing sky as many different shades.

R-e/p: You're doing the shading of sound with effects, level, and EQ?

JS: I like to use different types of echo. I might use a very short delay of the Lexicon 224X for a lot of the modern sessions. Natural chambers are good for some things. Some echo may be short chambers, long chambers, digital echo, discrete echo.

R-e/p: What do you mean by discrete echo?

JS: If an instrument is panned to one side, and echo put on it, the echo return comes up on the same side as the program. For instance: if the guitar is on the left, the echo comes back on just the left.

Even though I like the EMT 250, I'm not really that partial to the EMT 251. It does more, but I don't think it sounds as good as the original unit. It's pretty common to find that new pieces of equipment have more features than the original models, yet sometimes I feel that they don't sound as good as their predecessors. Of course that's not absolutely true; for example, I love the new [Lexicon] Super Primetime. I prefer to go for the *sound* of something, rather than the features.

There are so many new pieces of equipment coming out these days that it's hard to keep on top of all of them. I can't experiment during a client's session. Unless you're an engineer working at one studio full time, it's hard to get enough time to play with the new devices and discover what they'll do.

R-e/p: Do you use anything else besides space — echo or reverb — to shape the

“ It's very disappointing to hear something that's bitchin' on a set of big speakers, and then find out when you get home that it sounds like a pea rattling around in a tin can. Usually if you monitor anything loud enough, it sounds great; listen to it softly, and you really hear what's going on in the mix. ”

sounds of a mix?
JS: Simplicity — I don't think people like things that are complicated. I imagine somebody standing in front of a mirror, and pretending to be Michael Jackson, or whoever. I try to picture that when I do a mix.

As much as possible, I'm projecting myself into the place of the listener to get a feeling for what they want from the music. Engineering is a process of wearing a lot of hats; leaping backwards and forwards; being in the studio putting the whole thing together; and then being the listener at home. It's taken me a long time to be able to listen to something on a non-technical level — not passing judgement on a snare sound, or the amount of echo with an attitude that's too critical.

When you work on one project for a long time, you can get so inside of it that it's the "forest-from-the-trees" syndrome. But an engineer *must* get away from the project to look at the whole picture, and see how three-dimensional it is — to see whether what you have on tape is really what you're going to want when it gets down to vinyl.

R-e/p: If you're working with two or three engineers on a project, wouldn't it be easier to pick up that objectivity by tuning into the project once in a while, rather than being involved all the time?
JS: In some ways. Sometimes it's a little more of a challenge to work on somebody else's tracks, because you weren't involved with all the stages of putting it together, and now your job is to make all the parts work.

R-e/p: I would think you'd need very extensive notes and track sheets . . .
JS: . . . not really. The primary consideration is to write in fairly plain English. Then there's really no problem as long as the track sheets are accurate, and they describe anything that's happening with the individual program elements. "Track #16 is backing vocals in the B-Section and choruses," and that's really what it is — there's not a tuning note on it, or some old tambourine, or a bit of lead vocal that might have been comped from track #16 to a compiled track.

Or, quite often you'll put backing vocals on track #16, double them on #17, and triple them on #18, so you have three tracks singing the same parts. But you'll also get things like talking, laughing, or maybe somebody playing a tambourine or whatever in between vocal sections. To save the next engineer a lot of hassle later on, I'll go through the two-inch master tape before the mixing date, and erase all those extra noises. That way, once the tape is turned on, the next engineer doesn't have to worry about moving faders up and down to keep the mix clean. All the professional engineers I know have an overall respect for the next guy's work. We try to make it easier for each other. It's just courtesy.

Generally, instead of spending the time erasing unwanted sounds, we'll try to "tight punch" as much as possible. That means that if the vocals, for example, are coming in on a down beat, we'll punch in their track just before the down beat. Trying to clean up 10 tunes on an album, times 24 or 48 tracks, consumes a lot of time, not to mention being boring.

R-e/p: How do you prepare for those types of projects, where you "come in late," so to speak? Do you spend a lot of time listening to the existing tracks?
JS: Not really. I look at the track sheet, and listen to a rough mix to find out what comes in where. If you're good and experienced, you get an idea fairly fast of where the parts should be placed level-wise. In fact, if it's reasonably well-recorded, the tracks have a way of mixing themselves. A good arrangement with good playing and good sounds tells you where the tracks want to go.

I find that the mixing process either takes very little time — two of three hours, which I like, because I don't get burned out on the song — or a lot of time, like 10 hours.

R-e/p: What keeps you in there so long?
JS: You can't be "on" every day. Or, you might be working with someone who doesn't necessarily know what they want. A lot of what I do is a mental game of experimentation. Maybe the person you're working with — a producer, a client, an artist, whoever — thought they knew what they wanted within the first half hour. Yet nine hours down the road they aren't so sure, or what they *thought* they wanted was not what they wanted at all.

Other times, you have to go back to

the rough mix, because you just can't get that magic happening during the final. Technically, the [rough] mix may not be completely clean; there may be tracks that are left in, or background vocals that might have nothing happening for 16 or 18 bars, so you get tape hiss and such. But all that doesn't really matter if the *feeling* is there. Sometimes you nab that feeling; the majority of the time you get very close to it; and sometimes it doesn't work at all.

R-e/p: Do you start off your mix by relying on some standard or basic pan positions, and then fine-tune from there?
JS: Pretty much. "Stereo" is really a question of level from left to right and in-between. Basically, there are five positions in stereo: left, half-left, center, half-right, and right. By switching in a panpot, you put a mono signal, like a bass, anywhere between hard-left and hard-right. The bass usually appears in the direct center of the stereo image. Bass drum is the same. The rest of the drum panning depends on the perspective that the engineer prefers. Some engineers like a drummer's perspective; I like an audience perspective — sitting in front of a drummer.

Therefore, if I was in front of a right-handed drummer, the high-hat could be half- or full-right. The floor tom goes on the left; snare in the middle or slightly off-axis of the middle. Actually, the snare plays such an important role rhythmically that if it's one-sided, it sounds quite awkward. Plus, you want the power of the backbeat coming directly up the middle.

Stereo guitars and keyboards are usually split hard-left and -right. Background vocals get split hard-left and -right, or slightly off-center left and right, so they sit behind and to either side of the lead vocal. Percussion can be put where ever it works.

You have to remember that the center image of the stereo comes up 3 dB when you flip the mix to mono. In other words, any program material that is panned directly to the center, like the bass, vocal

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We look forward to a continuation of our new-found relationship with all of you.

Again, a sincere "Thank You",

AUDIO ANALYSTS

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or bass drum, comes up 3 dB in level, when collapsed to mono. On the other hand, the outsides tend to lose level. So it's always a good idea to check your mix in mono, and maybe bring the outside tracks up a bit or move them in towards the center slightly.

I heard of one studio that was equipped with a small, low-power AM transmitter. After you did a mix in the control room, you could go to your car, tune to that band, and see what the mix sounded like on the radio. I thought that was a great idea.

R-e/p: The George Benson album listed you and Michael O'Reilly as being the remix engineers on the project. Were both of you in the studio at the same time?

JS: Yes, Michael has assisted me with all of the work that I've done at Atlantic. I mixed the large majority of the Benson album with Michael's and Arif's help.

R-e/p: Did you each fall into any specific roles during the mixdown stage?

JS: We naturally find our own jobs. I try not to step on a producer's toes, or the assistant's toes; the assistant tries to stay out of my way. The idea is to have respect for and from someone that's working for you, or with you. And you have to *earn* it. Usually, if I think it's required, I'll make a comment that leans toward production. Or, Arif will ask me what I think of something. I'm as honest as I can possible be without sounding negative — you can't *always* love everything that's going on. It's teamwork.

And just as the producer leans on me, I lean as well. I don't know every second of a date whether something is absolutely right. So I'll bounce off a second: "What do you think of that?" "Do you think that sounds okay?" "Is that in the right place?" "What do you think of the echo? The placement?"

I don't believe that only *one* person is solely responsible for the finished musical product. There's the input from musicians; the sounds that they give me to work with are so important, because nobody can make a silk purse out of a sow's ear. [Session bassist] Nathan East always tells me that I get a great bass sound with him. Well, I'm flattered, but I'd have to be deaf to not get a good sound with him. The guy has a *great* sounding instrument, and is a phenomenal player.

There are so many variables — the producers, engineers, players, assistants, the studio, equipment, and so on. A record is all that and more working in tandem. Within certain parameters, we all have our "jobs" to do. But it's just a lot more fun — and quite often the results are a lot better — if everyone contributes whatever they can. If the assistant wants to lean over the console and adjust the balance on something, why should I get upset? An engineer *never* stops learning. And you always have to be aware of the fact that the

assistant, who is primarily setting up mikes, and making sure that the session runs smoothly from the studio's standpoint, in six months may be doing what you're doing, and blowing people away with his talent. You can't be paranoid about that, but you must realize the possibility.

R-e/p: You also mentioned that you often have to work in several different studios to finish one project. I would think that working in the same studio for an entire project makes more sense, in terms of maintaining a continuity, because you can get used to the sound of the studio, the echo, the assistant, etc.

JS: From my experience that way of working makes much more sense; I would rather work in one facility. But, being an independent engineer in Los Angeles, there are dozens of studios in town that I may have to work in. Quite often artists and producers and record companies need a single mixed in two days. You can't do a string ovedub and mix the single at the same time, so you must have more than one studio, and a couple of engineers.

In addition, the individual cuts may call for totally different ambience levels, so I never lock myself into one studio. One room may be perfect for a particular drum sound, another for an orchestra, and so on. Experience tells you the parameters of the room — this room is a bit shy on the top-end; this one has too much bass; this studio has a certain complement of mikes; this other one doesn't; and so on. You don't buy all your clothes at the same store, do you?

An engineer has to be able to walk into any facility and make it happen. That's what he's *hired* for, and that's what professionalism means. There's one thing that the best equipment can't get for you, and that's good sounds. You have to know in your head what you're going for. It's experience, knowledge, and having a good ear, as well as being in tune with the music and with what the artist wants to hear.

R-e/p: Do you have final say in the selection of which studios you'll do the project in?

JS: Pretty much. In the early part of January, Arif is scheduled to do the project with Phil [Collins] for a film. I think Music Grinder would be the ideal room for that. I know Phil, and the sound he likes, which is an ambient drum sound. Obviously, I can't get that working in a room that's 10-foot by 12-foot.

This facility [Music Grinder] is not overly expensive, and that aspect has to be considered, too. It also sounds great. People have far more limited budgets these days.

R-e/p: You've mentioned that you prefer to stay with a few select clients, but if you were going into the studio for the first time with an artist, what kind of homework would you do to prepare for the sessions?

JS: I normally buy two or three of the artist's previous albums, which tell me many things: what the artist sounds like, if I don't already know. But, if I'm an integral part of the music business, I *should* know. I try to find out who the engineer was and whether or not I know his work. But I don't try to assimilate anyone's style; I do what I do, and try to make it suit that particular project. Obviously, it's not a good idea to make a George Benson album sound like The Who, with a huge ambient drum sound and power guitars. It's what suits the music. So the albums give me a fair idea about the type of material we'll be doing, and I try to work within that framework.

R-e/p: The music of most artists doesn't really change that much from album to album, does it?

JS: It depends. Sometimes the changes are quite drastic. For example, Melissa Manchester has taken a slightly more modern direction over the last couple of projects. I think she's happy doing that.

Generally, the path is fairly well laid out in terms of what will and will not work. But sometimes what you didn't think would work is just what the album needs; I don't think there's been one project where I've preconceived my total approach to the recording process.

R-e/p: Do you meet the artist and producer the first day in the studio?

JS: The artist, yes. If it's a producer that I've not worked with before, I usually like to meet with him beforehand. He may play me demos of the material, and we get to know each other as individuals, because we have to spend a lot of hours with each other to complete a project that we'll all be happy with.

And it's *very* important that the artist and producer are happy not only with the musical aspect of the project but also with the way it sounds. If not, they won't be back. I don't ever like to work with an artist or producer just once. It's a little bit of a letdown to me if that happens, because the first project is a learning experience of getting used to them as co-workers and friends — learning how they tick. After getting in sync with them, you can capitalize on that during the next project. Hopefully, the next time you work together, it can only get better. ■■■

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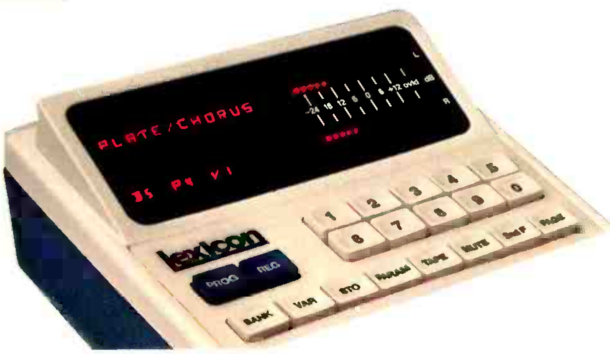

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For additional information circle #31

Carlo Sound System for the Oak Ridge Boys

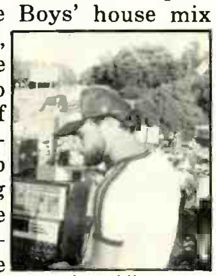
All Photography by DAVID SCHEIRMAN

Featuring the CS-3 Composite Speaker Cabinet

by David Scheirman

The Oak Ridge Boys are no strangers to "The Road." The award-winning vocal group travels to as many as 200 concert appearances yearly in its customized Silver Eagle coaches, just as they did over 10 years ago when the group was known as one of the hardest-working gospel groups in the Southeast. Since switching over to country in the mid-Seventies, the "Oaks" have won a string of impressive honors, including Vocal Group of the Year awards from both the Country Music Association and the Academy of Country Music, Single of the Year, Album of the Year, and Country Music Group of the Year.

Able to pick and choose from the many available concert sound companies, the Oak Ridge Boys' house mix engineer, John Mir, explains why the group uses Carlo Sound, Inc., of Nashville, Tennessee. "When the group first started doing major venues, we tried a lot of different systems," he recalls. "We have always had a rather hectic travel schedule . . . long overnight drives, and so forth, so reliability was a big factor to consider. And quick setup. We used to carry our own PA system right under the bus, in the luggage bay, many years ago. But of course that changed.



John Mir

"When we heard that Carlo Sound had a new all-in-one cabinet which still offered the benefits of horn-loading, we thought it would make a good system to fit our needs. We first picked the system up in the summer of 1980, I think it was, and we have used them ever since."

Carlo Sound first started doing business in the Nashville area in the late Sixties, when Rich Carpenter and John Logan leased their local band's PA system to the Johnny Cash Show. Other early supporters of the growing company included the Allman Brothers Band, B.J. Thomas, Rare Earth, Jackson Browne, and the Eagles. On Christmas Eve of 1980, disaster struck: a fire gutted the company's shop and office structure, destroying three complete road systems, including new Harrison consoles. "If it had not been for the large system we had out with the Oak Ridge Boys, we would have lost everything," Rich Carpenter remembers.

The sound company moved to new quarters near the Country Music Hall of Fame, and now offers a complete retail showroom and service and repair facility, as well as continuing to provide the Oak Ridge Boys with a touring sound system.

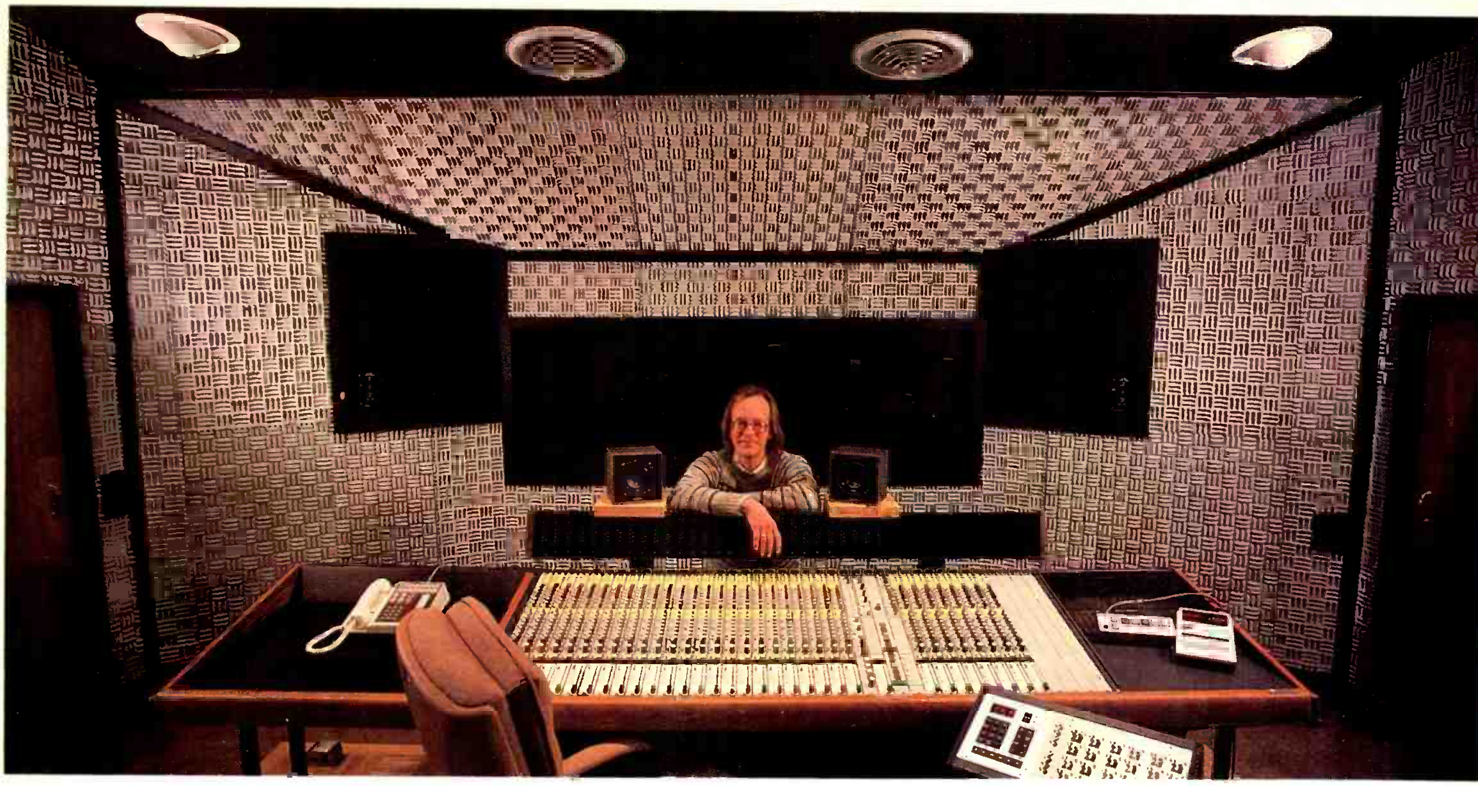
CS-3 House Speaker System

In 1978, Carlo began experimenting with composite speaker box designs, in search of the right compromise between a single three-way cabinet and large, horn-loaded bass and mid-bass sections, for which the company's systems were noted.

"We chose Eastern Acoustic Works to do the final cabinet assembly," Carpenter explains. "Basically, we took the EAW B215 bottom, with a few changes. I had a particular idea of what I wanted to hear in the mid-bass, and I made a model of a horn flare and sent it to Ken Forsythe. He played around with the design, added a phase plug, and then we integrated the Community Light & Sound Radical Radial fiberglass high-frequency horn . . . and came up with the CS-3: the Carlo Sound three-way integrated loudspeaker box."

The CS-3 cabinet stands an imposing 6½ feet tall on its casters (Figure 1). Each box weighs in at 422 pounds when loaded with two JBL 2225 15-inch speakers, a single JBL 2202 12-inch cone, and a JBL 2445 high-frequency compression driver. (This box is comparable in size to contemporary all-in-one cabinets in use by several major sound reinforcement companies which are smaller, yet lack the advantages of horn-loading in the bass and mid-bass sections.)

Additionally, CS-2 cabinets exist, which combine the bass and high-frequency components in a smaller, two-way box for additional coverage of full frequencies when needed. The CS-2



Jimmy Tarbutton, Chief Engineer

'Come On In'

Valley Audio invites you inside ACORN SOUND RECORDERS' new control room.

When the Oak Ridge Boys wanted a new control room for their Acorn Sound Recorders in Hendersonville, Tn., they entrusted their chief engineer, Jimmy Tarbutton, with the responsibility of contracting the best services available for the job. He chose Bob Todrank and Valley Audio.

"I wanted the latest in control room technology with a large functional space. Since we were building from the ground up, it had to be right. I chose Bob to completely design the room and oversee the construction. I wanted Valley Audio's technical services to do our equipment interface because of their more than ten years' experience in audio installations, and selected the new Harrison MR-4 32-24 console based on its flexibility and innovative design. We then selected a long term associate, Jim Aanderud of Viking Enterprises as our contractor."

The Oak Ridge Boys

Rear wall showing diffusion/reflection

Todrank says, "Since Jimmy wanted a large, open room with a very "live" feel, I designed a control room incorporating the latest *LEDE (Live End/Dead End) concepts. I chose a rear wall diffuser system designed by Peter D'Antonio of RPG Diffusor Systems, Inc., to accomplish a widely dispersed sound field around the console. We built and installed the very first of its kind anywhere and I was thrilled with the results. I also used our TECRON TEF equipment to place the final room interior treatments. The proper implementation of the LEDE design theory along with the use of on-axis monitoring, correct room geometry and accoustical equalization (selective diffusion/reflection/absorption techniques) has resulted in a room I'm very proud of."

The Oaks are proud of it too. Duane Allen's reaction... *"It's like a dream come true."*

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VALLEY AUDIO

2821 Erica Place • P.O. Box 40743 Nashville, TN • 37204-3111 • 615-383-4732 cabinets stack easily on top of the CS-3s, with the radial horns coupling for increased directivity of the system's high-end (Figure 2).

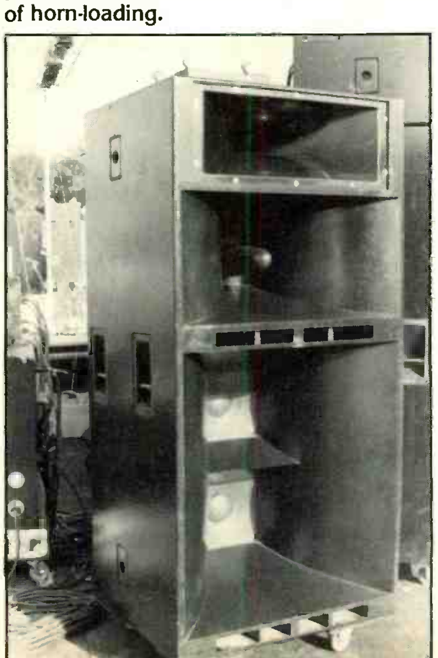
"For the Oaks, we usually carry 32 cabinets . . . 16 per side," states Carlo Sound's Steve "Gyro" Garrard. "We do an almost unbelievable assortment of venues . . . everything from State Fairs and auction barns, to Las Vegas and Atlantic City. We don't always use them all, of course, but when we do that gives us 64 15-inch drivers, all horn-loaded . . . plenty of system for what we need."

The CS-3 features permanently-attached heavy-duty castors, which are mounted to the bottom of the box with a ⅛-inch gasket made of neoprene foam, to prevent rattling of the wheels. "We found that to be necessary, especially when the system is flown," Carpenter points out. "The boxes are tuned to 40 Hz, and when we fly the system, one row goes upside down, so that the bass sections couple. The system puts out a tremendous amount of bottom end . . . but the wheels don't rattle!"

Hanging straps are attached directly to steel plates that are flanged and mounted from the inside, on 18-ply birch. Each plate is held on by 14 wood screws and glue. Handles are *not* load-bearing points, but are used strictly for ease of moving the cabinets around.

The back of each cabinet houses two multipin connectors, for receiving signal from the amp racks, and jumpering the boxes together in pairs. Additionally, a hook-and-loop strain-relief assembly enables the crew to attach each speaker cable directly to the box for positive connection, ensuring longer cable-connector life (Figure 3).

Figure 1: The Carlo Sound CS-3 loudspeaker cabinet, combining ease of transport in a composite box with the efficiency of horn-loading.



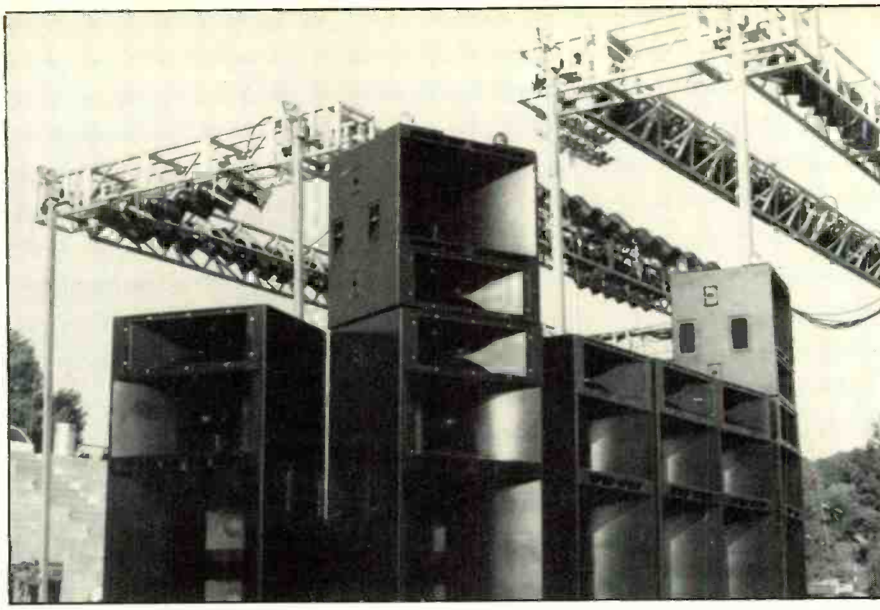


Figure 2: Speaker stacks at Southwest College in San Diego: six CS-3s per side, with CS-2 two-way boxes on top.

Amplifier Racks

Carlo Sound has been using Phase Linear amplifiers for 10 years. The current system is set up with Model 400 Series Two units, packaged eight to a double-wide rack, which comprises a wooden case that rides in a protective road case. Each individual amplifier is shock-mounted with strips of neoprene foam, and the tie-down screws are backed with rubber strips (Figure 4).

"One of the problems with the early Phase amps was their tendency to fall apart after a few thousand miles in the trucks," remarks Carlo engineer Mark Hunt, who mixes monitors for the Oak Ridge Boys. "With this type of mounting system, every amplifier is cushioned from the rack, and the rack is cushioned within the case. We don't have amplifiers fall apart anymore."

The CS-3 system is set up with a pair of 15s being driven by an amp channel, two 12-inch mid-bass sections on one channel, and two pair of high-frequency horns driven from one channel, at a 4-ohm load. House amp racks are equipped with a ¼-inch jack patchpanel in front for easy access should switching of signals and channels be necessary; the rack wiring is normalled to accommodate the CS-3 speaker system. Military-type multipin screw connectors on each rack accept the standard CS-3 cables.

System Front End

An early supporter of the Harrison Alive Series of portable consoles, Carlo still uses the board in a 32-channel configuration (Figure 5). Equipped with VCA grouping, LED ladder meter displays, and multiple mix output sends, Carlo engineers often find themselves having to give a crash-course in VCA mixing to opening-act engineers.

Cut to 10 minutes before showtime:

Carlo Sound Engineer: "Do you have any questions about the console?"

Opening Act's Soundman: "These are the submixes, or are they the matrix out?"

CSE: "Well, not really either one. They're VCA-grouping controls. They are not submasters per se."

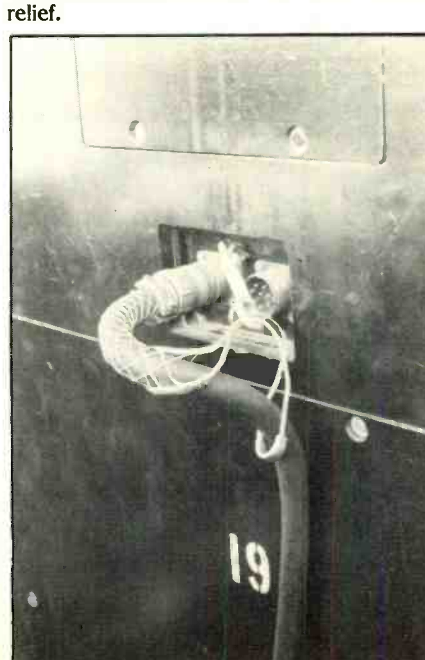
OAS: "Well, what do they do?"

CSE: "They control whatever groups of audio signals you assign to them. They operate voltage-controlled amplifiers."

OAS: "Yeah, but how come they're labeled vocals, guitars, drums, and so on, if they aren't submasters?"

. . . continued overleaf —

Figure 3: Back of the CS-3 two multipin connectors, and a hook for cable strain-relief.



built especially for us by the factory. We use it just to boost the system gain a bit, and to match up the board outputs with the other signal-processing gear."

From the line drivers, output signals pass through White Series 3000 filter sets, with 1/6-octave bass controls, and dbx Model 160X compressor-limiters. The signal was split with a Yamaha F-1030 electronic crossover. Three each of the crossover, compressor, and equalizer units were available. "We have a spare on-line for each unit," Garrard continues. "Or, it can be used as a third main output for a central overhead flying cluster, when we hang the system."

A Crown RTA-2 real-time analyzer completed the drive rack, fed with miniature Countryman pressure-sensitive, piezoelectric microphone. "This little mike works well in this application," he comments. "It's not calibrated to the RTA or anything, but it does check out to be pretty accurate."

The secondary house rack contained a Lexicon Prime Time digital delay; a set of six dbx 160X compressor-limiters (channel inserted on drums and bass); two Omnicraft GT-4 noisegates (for keyboards); an Aiwa 6900 stereo cassette deck for playback; an Eventide 1745M digital delay unit; and an URSA Major Space Station. The rack was equipped with a complete ¼-inch patchbay. Although not always used for the Oak Ridge Boys show, an EXR Exciter was also available.

Monitor System

Carlo's Mark Hunt mixes monitors for the show. His stage monitoring system is also centered around a Harrison Alive console. "This board is *great* for the stage," he says. "I'm using the Alive in stock form, with very few modifications. One thing we have done is to take the high-pass in/out switch on each mix output, and change it over so that it is now a Mix Kill switch, which comes in handy."

For the Oaks, Hunt uses nine of the Harrison's 12 available outputs. "Since this [board] is not really intended to be used for monitors, with a certain number of discrete, obvious outputs, I sometimes have to jockey around a bit to take care of opening acts. Here, our guest act is Michael Murphy. They want five or six mixes, which is no real problem . . . I have the open outputs. But for me to have to show another engineer who is not familiar with this console how to *get* the extra mixes might take some time."

One unusual aspect of Hunt's monitor mix layout was that a separate output mix had been provided on the downstage edge for each of the four vocalists. "They each have totally different ears, hear things differently . . . and they have different musical tastes," he explains. "Instead of going for one down-stage mix for everyone with a vocal blend, I sort of feature each singer stronger in his own box. The boxes are fairly close together, and the guys work sometimes 10 or more feet back from them, so it comes out almost as if there were four different amplified voices lined up in a row in front of them. But it works well."

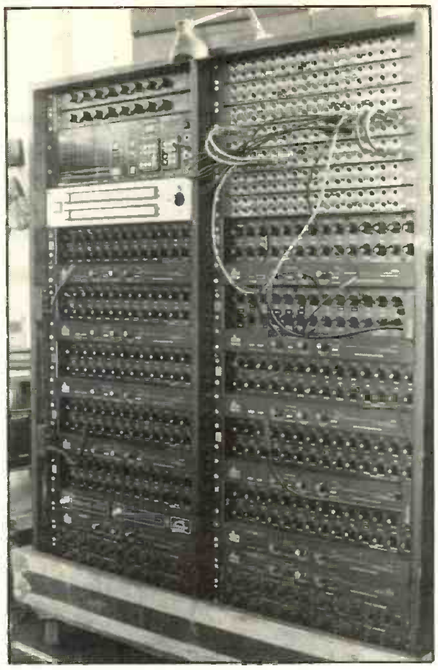


Figure 6: Monitor electronics rack with White filter sets for each output mix, and complete patching access.

Other mixes include one each for the drummer, saxophonist, keyboardist, guitarist, and bassist. "This group worked on so many small, crummy PAs in their gospel-group days, where sometimes they all had to share one monitor, that they are really amazed that we can set it up so that they can each hear whatever they want," Hunt comments. "The singers are fairly demanding . . . Duane [Allen] has a broad range, and does both low and high parts. Joe [Bonsall] sings the high parts — he has a lot of 4K in his voice, and needs a lot of that in his monitor.

"I would like to have the four monitors downstage all be EQed the same, but it just isn't possible, due to their different needs. Sometimes, if I pull out too much of one frequency for one of the guys, that EQ setting will mask one of the other guy's voice.

"Another advantage of having the four separate mixes is that when one of the guys has a solo on a tune, and is walking across the whole stage, I can follow him around as he walks, from box to box, without bringing up the level all over the stage on his voice, and blowing everyone else away."

Why no sidefills? "They have never liked sidefills," Hunt replies. "I think that, sometime in the past, they have been blasted away by them, and they are a bit gun-shy. I tired it once myself three years ago, but it just doesn't work well for us. To do it, I would have to have them set up in stereo, and be constantly panning back and forth . . . as one guy got too close to the left side, I'd have to pan him away from that box; that sort of thing. It would be pretty hectic.

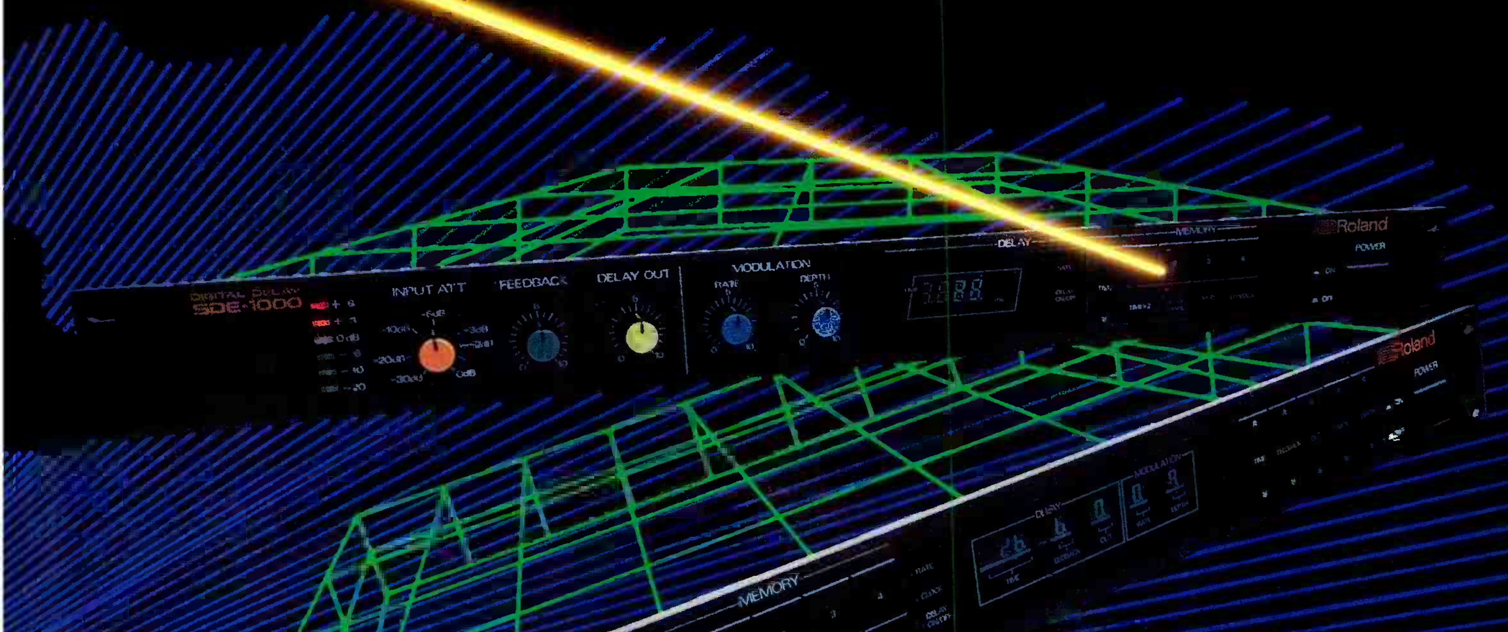
"Basically though, the singers have a mental block against sidefills. We could be having a problem with a room's acoustics, or something else, but they would look at the sidefills first as being the problem. So we just canned them."

Monitor electronics include White third-octave filter sets, as used in the house system. "Basically, on each output mix, I come out of the board, hit the EQ unit, maybe use a limiter, and then go to the crossover," Hunt explains. The system has six dbx 160X limiters available for patching onto output mixes. Crossovers for bi-amping the floor slants are by Phase Linear.

"The dbx unit has the Over-Easy fea-

Figure 7: Vocal floor slants comprised an EAW 2-12 cabinet loaded with JBL 2202 12-inch cones, and a 2441 compression driver.

INSTANT



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DELAY

OAK RIDGE BOYS

oned above the drum kit for use as an overhead (Figure 9).

Instrument amplifiers were given Shure SM-57s mounted on shock-mounted stands, and the acoustic piano received a Helpinstill pickup. An HME wireless unit served to amplify the saxophone, while an old Shure "bullet" mike was used when the reed man played harmonica.

And the vocal mikes? "The Oaks are doing an endorsement for AKG, so we always have different models to try out," Hunt points out. "Right now, I am using the Model D330BT for the guys, which has a switchable high boost at 7 kHz, and a low-end rolloff switch. It is a hypercardioid design, and I think they work particularly well on stage, due to their good feedback rejection characteristics."

On The Road With The System

"We have tried to get this system tuned up, and all put together, so that we have as little on-the-road repairs as possible," Hunt offers. "I do seem to end up doing a lot of soldering . . . cables are probably the weakest link in any system — mike cables, especially; these guys always twirl them around, that sort of thing.

"The main thing about touring with a system night after night is learning all of the things which *might* go wrong. After you have done it long enough, they will probably all happen to you at one time or another — feedback when you don't know why; blown power supplies; rain damage; dropped mikes. It will *all* happen to you in the course of a long tour."

One small touch that makes day-to-day setups and teardowns easier for the Carlo crew was the provision of Velcro tie-tabs at the ends of long speaker and microphone cables (Figure 10). As a cable is coiled, the tie-tab quickly and positively secures the wrapped cable bundle, doing away with the need for strings, tape, or worse . . . knots.

Also, monitor speaker cables are cleaned up considerably through the use of output distribution boxes; the low and high signal for each mix are carried down multipair 'speaker cable, and a break-out box with male connectors

DEVELOPMENT OF CS-3 LOUDSPEAKER CABINET
Cooperative design by Carlo Sound and Eastern Acoustics Works

Manufacturing firms often take an active role in the development of new products for use by touring sound companies. By way of an example of such cooperation, the CS-3 loudspeaker cabinet was a joint-effort project for Carlo Sound and Eastern Acoustics Works, initial prototype design being carried out at Carlo's Nashville plant, and subsequent cabinet design completed by EAW, who then produced the box for Carlo's exclusive use.

"Carlo first contacted us to get one of our double-15 cabinets for experimental purposes," recalls EAW's Ken Berger. "We sent that down, and Rich Carpenter and John Logan were impressed not only with the cabinet's performance, but also with our woodworking abilities. We use a number of proprietary construction techniques in our cabinetry, including a polyurethane-reinforced horn section, which allows us to execute very complex horns that exhibit both structural integrity and correct mathematics. So their decision was to have us fabricate the cabinets here at our shop."

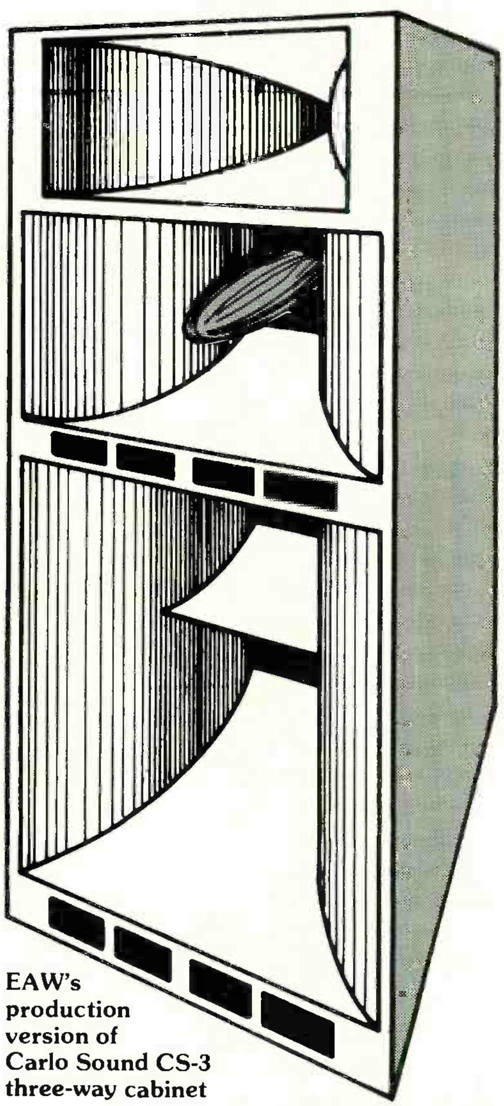
The CS-3's mid-bass horn section began with a Carlo prototype that happened to be amazingly close in concept and design to an EAW product nearing completion. "Carpenter sent up a plywood mockup prototype," Berger remembers. "And it so happened that we were simultaneously approaching the final design stage for our MR-102 midbass horn, which also used a single 12-inch speaker. The piece they sent up to us was just an approximation of the finished device, but it was right on the same track. We combined their ideas with ours, and built the mid-bass section onto the bass horn section. Space for the high-frequency horn was added, and we had the CS-3."

Unfortunately, most of the new CS-3 boxes were in Carlo's Nashville facility when a holiday season fire gutted the building. "The decision was made to go ahead and rebuild *another* complete set of 24 new cabinets for the Oak Ridge Boys, with whom Carlo had contracted dates that next month," Berger states. "That was quite a test of our production facilities. We had only two weeks to fabricate them from scratch, load the components, and ship them out to Seattle for the first date.

"Our biggest problem was plywood. Since we use Baltic birch in our products, it is not something that you can buy at any lumberyard. We always have some in stock, but the CS-3 cabinets require a specially-ordered five- by six-foot plywood panel, as opposed to the regular five- by five-foot sheets which we stock. Our importer was able to find some here in this country, and get it to us . . . otherwise, we would have been waiting for another month or so for the boat to come over.

"But they all got built, the Carlo crew flew up here to supervise component loading and connector wiring, and our shipping agent made a special weekend pickup when he heard that it was a 10,000-pound freight shipment!"

Currently, Eastern Acoustics Works offers three one-box, horn-loaded off-the-shelf loudspeaker systems for sound reinforcement. "Our prime concern is the building and marketing of our own line," stresses Berger. "However, we also like engineering. We have taken on numerous special design projects, such as the Carlo CS-3, and will continue to do so. We have designed special subwoofers which are used by NASA labs, and done projects for Bose, Gauss, dbx, and Richard Long Associates. And we love helping concert sound companies!"



EAW's production version of Carlo Sound CS-3 three-way cabinet

Quoted specifications of EAW's off-the-shelf CS-3, which can be supplied in versions to accept JBL, EAW, or Gauss drivers, include frequency responses 40 Hz to 10 kHz, -6 dB, 80 Hz to 12 kHz, ±3 dB; HF sensitivity 107 dB 1W at 1 meter; MF 106 dB; LF 108 dB; HF power handling 40W RMS, 80W program; MF power 150W RMS; LF power 300W RMS; recommended crossover of LF/MF: 300 Hz, and MF/HF: 1.2 kHz, with a slope of 18 dB per octave; and a recommended high pass filter of 40 Hz, 18 dB per octave. □□□

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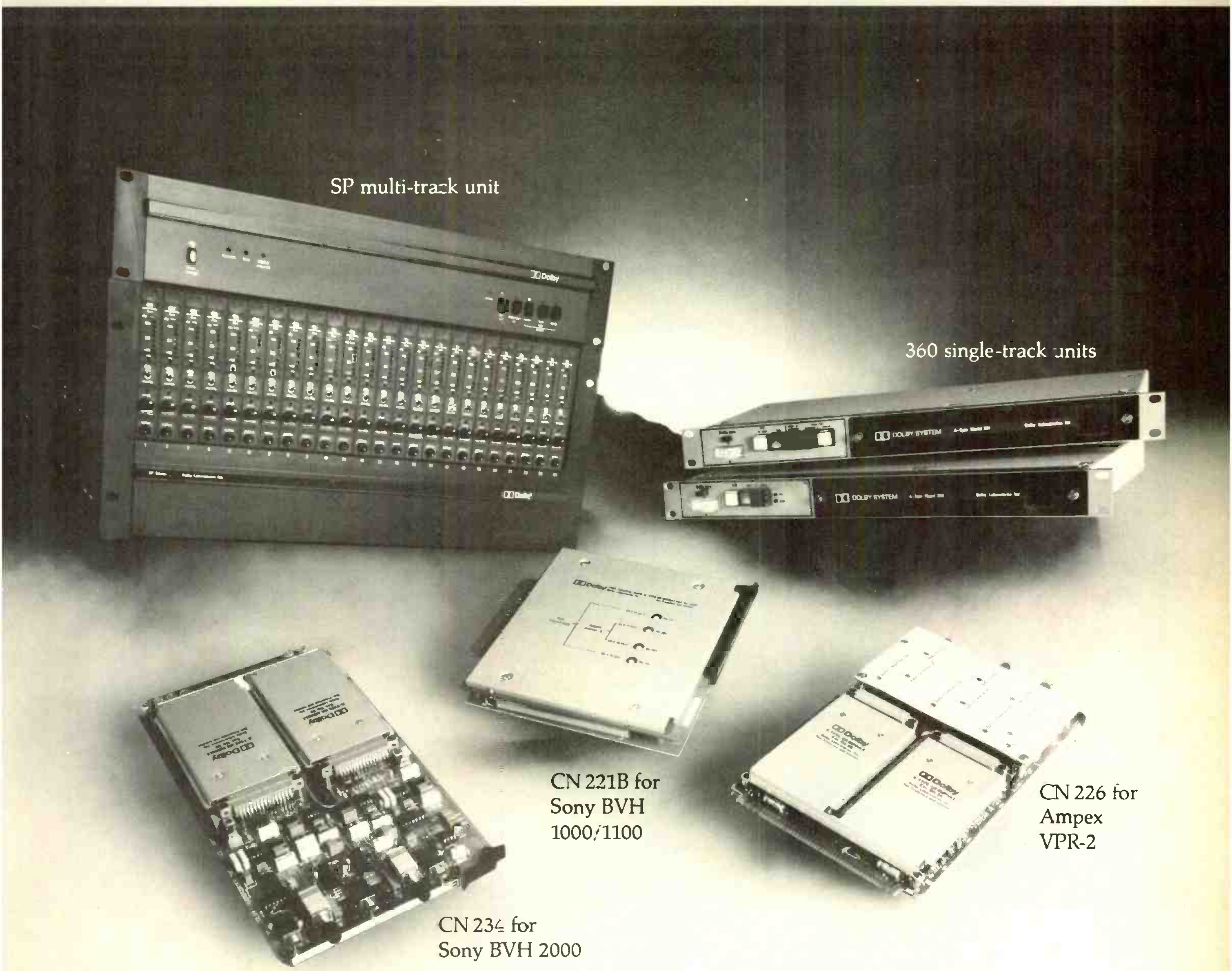

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SOUNDMAN'S NOTES FROM THE ROAD

Andy Chappel's Venue Notes from Across the Country

This issue marks the beginning of a new venue column culled from the notes of live-sound engineer Andy Chappel, who has handled all aspects of live sound for a variety of artists, in a multitude of arenas and clubs around the country. Currently on the road doing house sound for The Motel's tour, Andy has also worked with Marilyn McCoo and Billy Davis Jr., Aretha Franklin, Del Shannon, The Dickies, Lords of the New Church, Quiet Riot, and "X."

While *R-e/p* hasn't the space available to cover full details of each venue, nor the personnel to keep up with the constant changes happening everywhere, we will supply the basic conditions and who to contact for more information.

Just remember, it's a jungle out there and if you see Andy tell him to phone home — next issue's column is due.

EAST SIDE CLUB
1229 Chestnut Street
Philadelphia, PA
(215) 564-3343

Type of Venue: Club
Capacity: 500
Andy's Rating: * *
Acoustics: Very Dead
House Soundman: Albert
Mixing Position: Center of club in back by bar.
Console: Tangent 24 × 4
House PA: CLS cabinets w/JBLs & Crown power amps.
Monitors: None
Stage: Small, 20 × 20 foot
Microphones: Shure, EV
House Power: 100 amps, 3 phase.
Crew: Good
Load In/Out: Hard. From small back alley, down stairs.
Overall View: Small club; not a place to bring in much more than monitors. PA sounds good.
Recommendations: Bring in monitors and maybe a house console.

METRON
400 S. Cameron Street
Harrisburg, PA
(717) 234-1316

Type of Venue: Club
Capacity: 600
Andy's Rating: * * *
Acoustics: Dead Room
House Soundman: Bob Reed
Mixing Position: House right
Console: Soundcraft 24 × 8
House PA: None, outside PA company; Bob Reed Sound (717) 854-8871.
Monitors: None
Stage: 30 × 15 foot.
House Power: Good. 200 amps, 3-phase.
Crew: Good. Stage manager, Dennis.
Load In/Out: Okay. Through side door, 7′ × 42″.
Overall View: Okay. Room is nice, PA sounded clean.
Recommendations: Advance gig with sound company to make sure they have everything you need, or just bring your own sound.

PARADISE
967 Commonwealth Avenue
Boston, MA
(617) 254-2054

Type of Venue: Club
Capacity: 560
Andy's Rating: * * *
Acoustics: Real Dead
House Soundman: Seth Geiger
Mixing Position: Center of house, up high.
Console: Tangent 24 × 1
House PA: 4-way w/18s, 12s, JBL mids, Renkus tweeters.
Monitors: Good, 1 × 15 cabinets, 4 mixes.
Stage: 32 × 18 foot.
Microphones: Shure, Beyer
House Power: 200 amps, 3-phase.
Crew: Good
Load In/Out: Okay; through double doors from alley.
Overall View: A nice place to work; PA sound good; plenty of level.
Recommendations: Bring in your own monitors if anyone needs sidefills and/or more mixes.

LIVING ROOM
273 Promenade Street
Providence, RI
(401) 521-2520

Type of Venue: Club
Capacity: 550
Andy's Rating: * * *
Acoustics: Good, wood walls, curtains around stage.
House Soundman: Paul Martin
Mixing Position: Center of house.
Console: Stevenson Interface 24 × 4.
House PA: Supplied as needed by C&M Sound (401) 521-2520; Klipsch boxes.
Monitors: Good; four mixes.
Stage: 32×18×3-feet high.
Microphones: Shure, Sennheiser
House Power: 100 amps, 3-phase.
Crew: Good
Load In/Out: Through double front doors, and onto stage.
Overall View: Good gig; everyone helpful; PA sounds good; PA company can get nearly anything you need.
Recommendations: Advance the gig; you might need more monitor mixes than usual.

TOAD'S PLACE
300 York Street
New Haven, CT
(203) 562-5694

Type of Venue: Club
Capacity: 650
Andy's Rating: * * *
Acoustics: Okay when the room is full.
House Soundman: Keith Dupke
Mixing Position: Center of house.
Console: APSI 24×4
House PA: Good stacked and flown system supplied by Horizon Sound (203) 562-5582.
Monitors: Four mixes.
Stage: 32- by 16-foot
Microphones: Sennheiser, Shure, & Beyer
House Power: 100 amp, 3-phase.
Crew: Good
Load In/Out: Okay from front door, off the street.
Overall View: Okay. House PA system is fine, no need to bring your own.
Recommendations: Advance gig to check monitors and/or order more mixes if necessary.

RITZ
119 E. 11th Street
New York, NY
(212) 228-8888

Type of Venue: Large Club
Capacity: 1,450
Andy's Rating: * * * *
Acoustics: Good
House Soundman: Charlie
Mixing Position: Center of house, rear.
Console: Yamaha PM2000 32 input.
House PA: Good system; JBLs powered by Crown amps.
Monitors: 4 mixes, okay unless band is loud.
Stage: 30 × 24 foot.
Microphones: Shure, Beyer, & Sennheiser.
House Power: 200 amps, 3-phase.
Crew: Good
Load In/Out: Up two flights of stairs.
Overall View: Good gig; load-in is rough, but everyone is helpful and crew is good; PA sounds fine.
Recommendations: Bring your own monitors, and everything will be fine. □□□

Note: Andy Chappel's personal rating scheme is based on a maximum score of four stars.

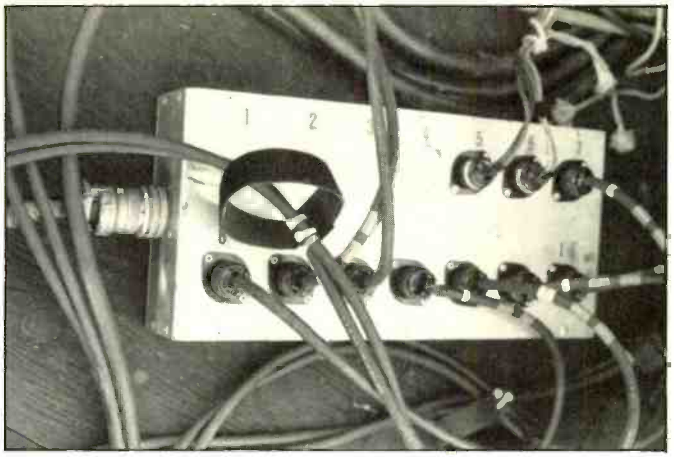


Figure 10: Monitor bi-amp stage cable distribution box, featuring a Velcro tie-tab for quick and easy cable storage.

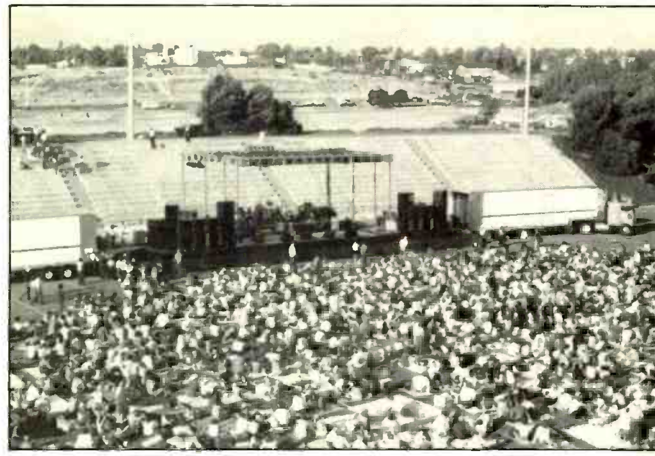


Figure 11: Carlo Sound system as set up outdoors, with six cabinets per side, plus two CS-2s on top.

enables short jumpers (with two different-sized connectors for lows and highs) to be run to each box. An 80-foot length on the multipair assures adequate reach on even the widest stages.

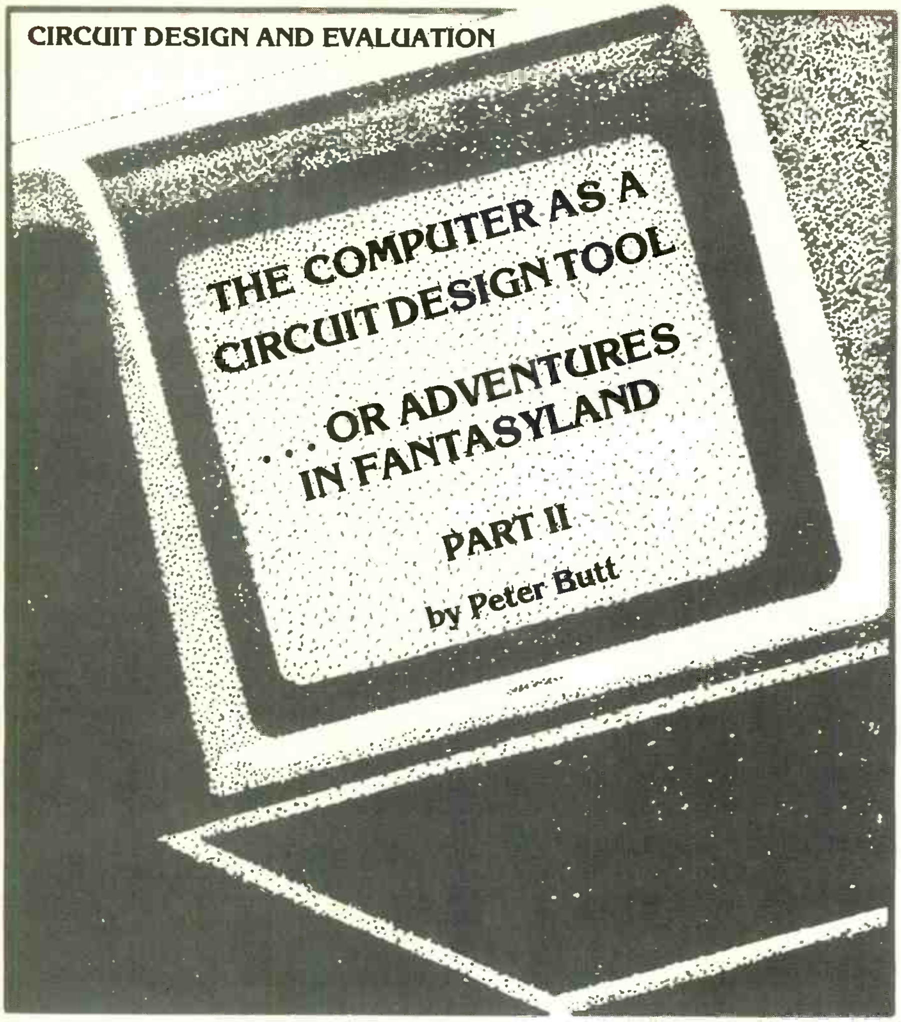
Sound Of The Show

The concert this writer observed took place at an outdoor amphitheater-type sports bowl at San Diego's Southwest college. The venue will seat approximately 10,000 patrons when set up with the stage on one side of the field.

House engineer Gyro Garrard and monitor mixer Mark Hunt had set up only six CS-3 cabinets per side, plus an additional pair of CS-2s on top of the speaker stack. The cabinets were arranged in a gentle arc outwards, with the backs of the cabinets in alignment. (Hmmm, I thought to myself, that's 28 15-inch cones . . . only one woofer for every 357 people, and outdoors, yet!)

The system's horn-loaded bass and mid-bass sections did an adequate job of making the sound of the group seem quite present, even at the top of the high concrete-tiered seating area (Figure 11). Certainly not enough acoustical energy to power a Heavy-Metal Battle-Of-The-Bands . . . but then, this was Country. And there were another 24 cabinets sitting in the semi.

It was refreshing to see a sound crew have the sense to not set up a tremendously large sound system when it wasn't needed, just because someone had paid for the diesel fuel to bring the PA to town. And, as diesel fuel becomes more expensive, perhaps hybrid systems such as the Carlo CS-3 loudspeaker units, which offer a unique compromise between the efficiency of horn-loading, and the ease of transport of the all-in-one cabinet, will become more popular.■



In part one of this article, published in the October issue, the author considered the various merits of a computer-aided circuit design system, and considered the real-world example of a digital anti-aliasing filter. Having determined the group-delay response, and established the importance of distinguishing between relative and absolute phase response, this concluding part moves on to look at the optimization of circuit delay times, time- and frequency-domain transformations, and circuit response using a digitized cymbal crash.

Variable Circuit Delay Time

One of the features of this program is that the COMTRAN Optimization process permits the circuit delay time to be one of the parameters allowed to "float" during iterations. The importance of this capability can be explained by first discussing the analogous case for a relative amplitude response optimization. Because it is generally not possible to predict the precise amplitude response that will be finally arrived at upon termination of the iterations, some kind of device must be included in the optimization process to allow the program to "trim" the over-all average circuit gain, so that it will not reject Relative system response which otherwise would be acceptable only because of Absolute gain deviations. This can be done by the user adding a resistive divider at the network output port — say about 20 dB-worth — and cranking the 20 dB loss into the target data amplitude values. The divider tap then is taken as the network output node. By allowing the program to trim the lower leg of the divider, variations in circuit gain can be compensated for and Absolute system gain then trimmed or optimized later if necessary.

For the same reason, it is not possible to predict the final result of the Absolute group delay resulting from a Relative phase-related optimization. If it is impossible for the group delay value to "float" with each iteration, the program will abandon the optimization process because the Relative phase corresponding to the iterative circuit values are inconsistent with a fixed value of Absolute group delay. The COMTRAN program permits selection of the group delay correction factor as one of the optimization parameters; without this feature it would not be possible to solve the kind of problem under consideration here. I think that the importance of such a capability will be apparent to anyone encountering the need to devise group delay correction networks for loud speaker crossovers, anti-aliasing filters, and frequency-division multiplex networks, or charge-coupled device filters, to mention only a few potential applications.

Awareness of the difficulty of this type of problem in human terms is necessary for one to appreciate the incredible power of this magnitude of numerical capability. The iteration circuit response points corresponding to the target data points are matched for a least-squares fit to the target data points, a least-squares approximation to the target data points desired being the result of this process.

Try doing that on your slide rule.

It has taken this writer about three months to gain what appreciation he has for this design and analysis tool. It is difficult to adequately convey the experiential impact of having such power available on a convenient and routine basis.

Time- and Frequency-Domain Transformations

The human need to gain some kind of intuitive insight into the consequences of a design effort has been attended with the inclusion of the AC-CAP companion program, S-WAVE, which permits the transformation of transfer function data generated by the AC-CAP Waveform Generator subprogram into time-domain data sets. As touched upon briefly earlier, the transfer function is configured as a set of data points, each consisting of a triplet of numerical values representing driving frequency, circuit gain at that frequency, and circuit Absolute phase response at that frequency. The transfer function may also be written as an equation whose variables are called "complex variables," because they represent quantities that can be expressed as the sum of a pure number (a "real"), and the product of a real number and another quantity that is the square-root of minus one (the "imaginary"). Numbers that are composed of a real and imaginary part are called complex numbers; similarly, variables in these complex numbers are referred to as complex variables.

The equation form of the transfer function does not readily occur in the context in which COMTRAN operates, but that is not a major difficulty, because the discrete data triplets alternatively convey the same information. One may not look like the other, but they are the same information. The data triplets could be derived from repeatedly solving the algebraic transfer function for discrete frequencies, and *vice versa*. Not necessarily within the scope of this program, though.

If a time-domain function in the form of, let's say, an ideal square wave is Fourier-transformed into its frequency components, each associated with an amplitude and phase value, and if these frequencies correspond to the array of frequencies used to describe a transfer function, it is possible to operate on these two sets of data in such a way as to

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cond, then the cymbal-crash data file, called "Cymbal," will convolute with it with no spectral shifting.

Figures 11 and 12 show the Cymbal waveform as digitized; Figure 11 is Cymbal after passage through the uncorrected anti-aliasing filter. It can be seen that the filter's effect is to minimize the asymmetry of the signal, reduce its peak-to-average ratio, and to slightly widen the intervals separating its zero-crossings. Although the corrected signal of Figure 12 still suffers from some broadening of zero-crossing time intervals, it is not quite as severely affected as is the direct filter output signal. Also it will be noted that some of the asymmetry of the original signal has been restored, and the tendency for the valleys between crests to be filled in, and for the peaks to be rounded off, is also less evident. The peak-to-average ratio also has been improved.

It should be noted that these waveforms are plotted with vertical scale factors such that the 6.6 dB filter loss has been accounted for in the overlays, so that amplitude values may be directly compared to the reference Cymbal trace which is another handy capability of S-WAVE, since it permits time and magnitude scale adjustment of successive traces to be plotted on the same graph for comparative purposes, such as gain and symmetry evaluations.

Figure 13 shows the uncorrected and corrected filter Cymbal output signals overlaid on identical amplitude scales with delay times shifted for direct comparison. The two traces do not match precisely in time because the group delay is not truly a constant through the pass-band, and some of the signal frequency components are still arriving at skewed times.

A 'scope photo of the corrected real filter 1 kHz square-wave is shown as Figure 14; note the similarities between this photograph, and the corrected plot of Figure 9. Such agreement with predicted corrected filter square-wave response instills confidence that there is, after all, some sort of corrolation between theory and practice.

Murphys' corollary to the contrary may be called into question here.

The reader must be aware that the whole optimization process did not take place in anywhere near the space of time it takes to read this article. It also must be emphasized that there are many operations done as checks on the validity of methods of solution, blind alleys, and tangential paths taken that vanity begs not be enumerated here. We all make mistakes . . . we all don't care to admit it. That fact is no cause for shame, provided we are careful enough to catch them before relying heavily on those errors, or before they are discovered by others and brought to our attention. It should be apparent to any experienced circuit designer that the method of solution described here could not be done in any reasonable period of time without the aid of the computer, and especially not without the use of a powerful program group, such as COMTRAN.

There are times when a perfect square-wave doesn't represent the kind of signal dynamics that the circuit will be seeing in real-life application. If desired, the rise- and fall-times of the stimulus square-wave can be modified by use of an S-WAVE utility called "Generate Stimulus." If that won't quite fill the bill, the stimulus waveform may be defined or modified manually as a series of time-domain vertical axis numerical values corresponding to horizontal axis time interval addresses. Stimulus waveforms also may be acquired from real-world sources, if there are appropriate instruments available that will accept HP-IB (IEEE-488) bus control. COMTRAN contains a

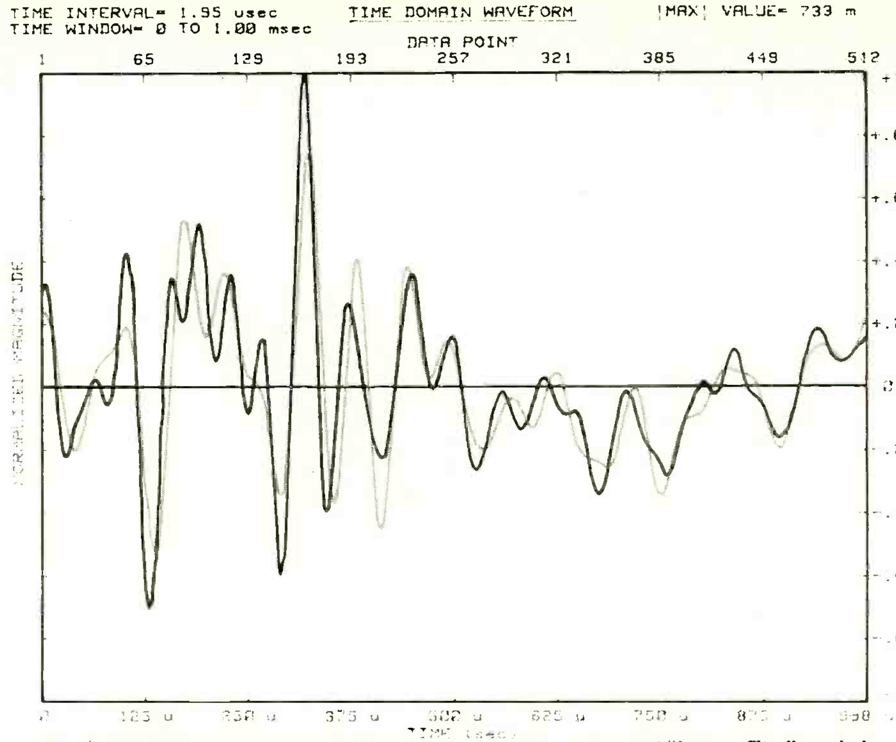

Figure 12: Reference cymbal waveform (grey), overlaid on corrected "fantasy filter" cymbal response. Scaled to compensate for 6.6 dB insertion loss.

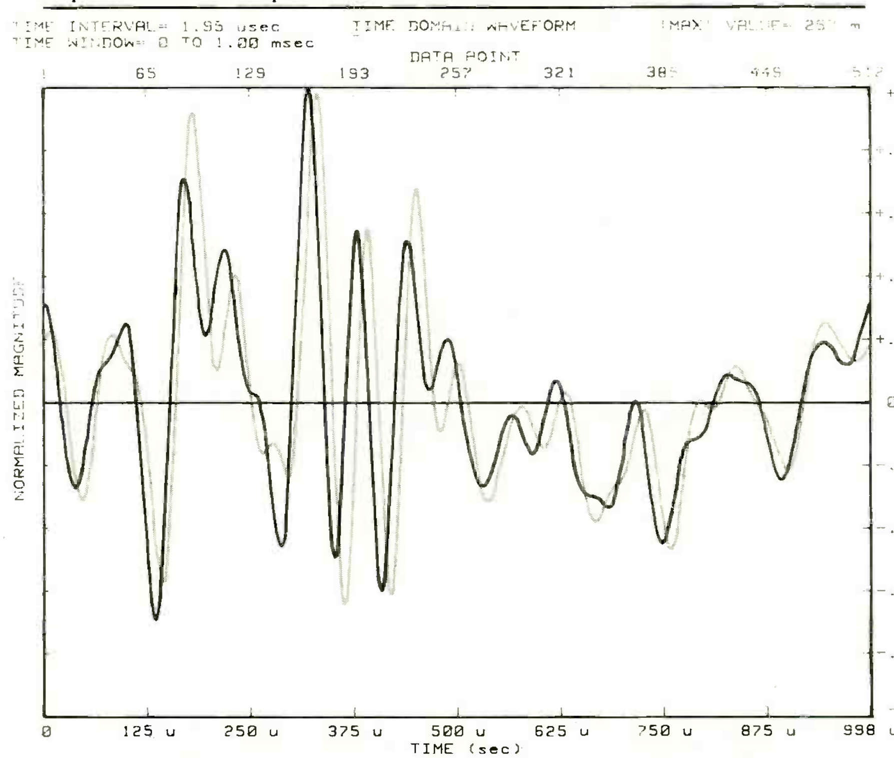

Figure 13: Response of uncorrected model filter to cymbal stimulus (grey), overlaid on trace of delay-corrected filter response.

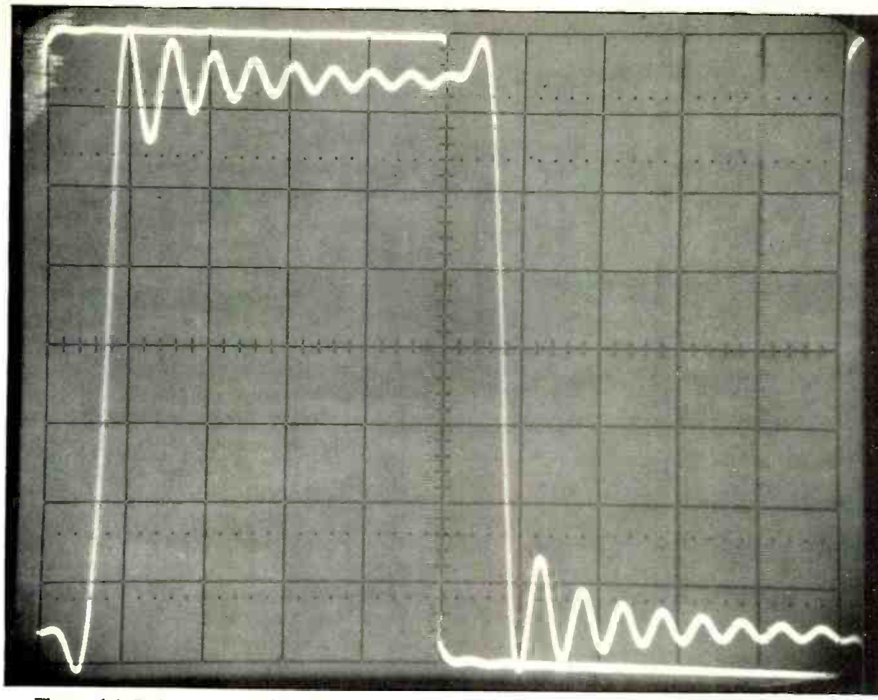

Figure 14: Delay-corrected 1 kHz square-wave response of real filter. Vertical scale: 0.5V per division. Horizontal scale: 0.1 millisecond per division (118.5 microsecond delay at half-height).

user-variable program that allows the computer to be programmed to drive and access instruments and acquire signal data that can serve as a transfer function driver, or will lend themselves to Fourier transformation, spectral analysis, and correction for sample interval error correction, and sample trigger-offset adjustment. Once in RAM, these files can be saved on mass storage for future use, or manipulation with other data files.

Other data-domain manipulations that can be performed as required on any data arrays fitting the S-WAVE data format are graphic display time-scale expansion by successive factors of two, and stimulus phase-shifting by successive -90 degree increments. Numerical integration of the driving function data file will result in a triangle driving function. Simply taking the inverse Fourier transform of the transfer function residing in the primary data file, will yield the time-domain function describing the system impulse response. Taking the inverse Fourier transform of transfer function in the primary data file, and performing integration on the result, will yield the system step-response in the time domain. Multiplication of a series of two or more transfer functions can be accomplished, two at a time in sequence, to yield the final transfer function that would result if the physical systems were connected in cascade, each with a perfect inter-stage buffer. Simulation of time-gated circuits, such as sample-and-hold circuits, or resettable integrators, can be achieved by use of another feature that allows setting desired parts of a data file to zero if that file has not been integrated, or to a constant numerical value if integration has been performed.

To fully understand and appreciate the potential of S-WAVE, it is helpful for the potential system user to have had some introduction to integral and differential calculus and complex variables. It is difficult to convey by word alone the impact that possession of such analytical power can have on one's perception of the physical world. Demonstration of real physical system response can be done in the space of an hour or less, without the need for instruments, parts, or soldering iron. Performance of existing systems can be analyzed without the need for access to that system itself. The veracity of published technology can be tested in as short a time, and the experiential significance of that technology absorbed by the user. Demonstration of academic concepts that are difficult to observe directly in the physical case become amenable to examination at leisure.

As a further advantage, there is no need to order parts.

The means for simulating the active elements of a circuit model have not been treated in any detail in this article, since an abundance of data on this subject has been published down the decades. While no need has been seen to discuss them in detail here, the following references are given for those interested in the subject. Vacuum tube models are discussed in reference #10. Treatments of bi-polar transistor models can be found in #11 and #12. MOSFET and JFET models are discussed in #13, while operational amplifiers are discussed in #14. A more detailed discussion of wideband operational amplifier models as well as high-

. . . continued overleaf —

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COMPUTERIZED CIRCUIT DESIGN

frequency bi-polar semiconductors is given in #15; a version of this op-amp model was used throughout the examples discussed in this article. COMTRAN constrains the user to model all of his active devices using current sources known as "Norton" sources or generators — these may be found described in #16. The Thevenin companion to the Norton source is discussed in #17 and #18. The choice of what kind of active element model is desirable for any purpose is up to the judgement of the program user.

A Solution Looking for Applications

Often in the course of daily ministrations to the recording industry I am overcome with the feeling that "normal" system performance somehow just isn't quite right. I'm sure that many of us have had that uneasy feeling that, even though this particular widget performs in the same way that all other known widgets do, that there is something that just doesn't ring as true as it might. Or, the frammis sounds funny when it's taking feed from the widget, or compression seems to limit the bandwidth; it sounds gritty on horns; etc., etc.

Armed with a widget and frammis schematic, adequate semiconductor data, and a set of programs like COMTRAN, it is possible to analyze the widget and the frammis in ways that would otherwise take, perhaps, days of measurement time, and days more for data reduction and analysis. I have discovered that venerated mainstays of the professional audio industry over many years have internal design problems, such as conditional stability, non-linear phase response, gain-limited equalizer circuits, current-limited line drivers, bias trap designs that impair signal transient and phase response, oscillation on clipping, non-optimal gain structure, widely-varying input/output impedance, and so on. Who knows what else is out there? I'm sure I'll find many more subcutaneous warts and a few tumors in the collective body electronic as time passes, and my facility with COMTRAN grows.

My experience with this program group is that its use promotes constantly greater sensitivity to the nuances of physical system composition and performance parameters. As has been the case for all of the powerful tools I have been able to acquire and use in my past work, my perception of reality is altered by the insight the tool has made possible. It's like a myopic suddenly discovering that eyeglasses can be cleaned.

I should not undertake to severely criticize existing designs of years gone by. I want to take this opportunity to apologize to any who might feel embarrassed by any of my closing comments. After all, those designers didn't have elegant circuit-analysis programs like COMTRAN. ■■■

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COMPUTER OPTIMIZATION OF AN EXISTING DESIGN: CASE STUDY OF A TRANSFORMERLESS MIKE PRE-AMPLIFIER

As a brief example of the power of COMTRAN software as an analysis tool, a transformerless microphone pre-amplifier was chosen for study. The circuit consists of the commonly encountered differential input instrumentation-type amplifier having low-noise monolithic bipolar transistors in its first gain stages that acted as impedance convertors for each side of the differential input. The differential signals are then summed by a summing amplifier, and fed to the outside world by a low-impedance driver circuit. The problem with this particular pre-amp design was that the transparency of the signal obtained from it was noticeably deficient as compared to the results from pre-amps of supposedly less *avant garde* design.

The first step in the analysis procedure was to obtain a schematic and parts list of the device, and then to reduce the schematic to an equivalent circuit involving voltage-dependent current sources and passive components. The overall system gain and phase characteristics for a gain of 51.84 dB are shown in Figure A. It will be noted that the amplitude response starts to fall off above 10 kHz, reducing to unity at about 3 MHz. The amplitude response is down about 2 dB at only 20 kHz, portending a muddy high-end, as has been confirmed subjectively.

The major portion of the pre-amplifier gain is derived from the action of each of its two differential input legs. Analysis of what might be going on in each of these identical stages was checked by moving the output node from the intended output port of the equivalent circuit model, to one of the final nodes prior to the differential summer input, which permits us to see the amplitude and phase response of a single differential input circuit prior to summing. The gain-determining components of these differential gain stages consists of a single variable resistor connected as the shunt portion of the feedback network of each of the opposing input circuits. Since the feedback loop includes the entire differential gain stage, we can get some inkling of the amplitude and phase response of a major portion of the pre-amp gain structure. This data are shown in Figure B. Note the early amplitude roll-off, as well as the "kink" in the phase response curve in the 50 to 100 kHz region. Although the phase lag does not cross the –180 degree line until about 3 MHz, indicating a somewhat stable closed-loop condition[1], the undulation of the phase response indicates a non-minimum phase condition in this circuit[2]. (Which is encouraging only in that it gives us something to chew on as we search for a remedy for the illness).

A number of small capacitors are used as phase-lead feedback components in each of the input stages, and also in the differential summer circuit. A relatively simple cure for the malais might be to remove these caps altogether as a sure-fire way to extend the system bandwidth. "A swell idea," we theorize. Let us set those caps to 10E-15 Farads, substantially deleting them from the circuit model.

TRANSFORMERLESS MIKE PRE-AMPLIFIER OPTIMIZATION — *continued . . .*

The results of this extreme nostrum are shown in the curves of Figure C. The amplitude response indicates a low-Q peaking effect prior to roll-off slightly below 1 MHz, while the phase lags to nearly −180 degrees at 10 MHz as the stage gain hangs in there at about +8 dB. Not a good omen, since oscillation of this stage is a distinct possibility under these conditions. The asymptotic behavior of the phase response suggests a bit of "brick-walling" might be going on as the circuit approaches its physical limitations at high frequencies.

Since our goal, by the way, is to achieve a significant improvement in the performance of this circuit, while leaving our clients' net worth substantially intact, we are limited in our alternatives.

Cheap is good, if it works; words to live by in any discipline. Our preliminaries seem to show us that there is definitely too much of some "thing" embodied in the circuit as originally configured. A dearth of that "thing" doesn't seem to be the ticket we are looking for, either. The question arises then: Is there a compromise between these two extremes that is livable, and also acceptably cheap? COMTRAN to the rescue once again, folks. By successively choosing decreasing values of the phase-lead capacitors, and checking the gain/phase curves for each case, we are able to get an idea if this approach has a chance of fulfilling our heartfelt desires. Figure D shows the results of a choice of value that seems promising. Stability is still a bit "iffy," but we just might be able to pull this one off with only a change of four capacitor values. Figure E shows the entire modified system gain and phase response, demonstrating that we are down 3 dB at about 78 kHz, as opposed to −3 at something less than 30 kHz for the benchmark.

Application of the Acid Square Wave Test, as a confirmation of either the cleverness or naiveté of our solution, is now in order. Generating the transfer function file for both the benchmark stock circuit and the proposed panacea, we then load the S-WAVE program and operate on the transfer functions to correct for the differing delay times of the respective circuits. A plot of the resulting square-wave responses is shown overlaidd in figure F; the slow one is obviously the original circuit response. The modified response is clearly faster.

The time domain plots taken for 512 data points showed traces of what might appear to be high-frequency instability in the form of low-level "grass" during the time slightly after the positive transition and slightly before the negative one, for both the new and old circuits.

To clear up this uncertainty — since unconditional circuit stability is a highly prized performance characteristic — a second transfer function over the same 1 millisecond time window was taken having 2048 data points for each case. The results of those samples were plotted for inspection; as can be seen in Figure F, there is, in fact, no trace of oscillation for either circuit.

In cases where the transfer function generated does not adequately include a significant portion of the system amplitude response, the lack of complete system transfer function data will result in information relavant to the time-domain response being unavailable to the program, and thus will result in a manifestation of this missing data. Mother Nature knows when something's wrong, and COMTRAN does, too. The missing part of the transfer function makes the square-wave response uneven, since the complete data are not available to fill in the rough spots. The bandwidth restriction of the smaller sample limited the amount of data we had to deal with, and therefore would have arrived at erroneous conclusions had we not gone further. The reader will recall that, although we are modeling an *analog* system, we are doing so *digitally*. This apparent incongruity indicates that Shannons' Law of information theory holds *inside* of computers as well as out.

Oscilloscope observation of the real modified pre-amp circuit shows no evidence of oscillation below 100 MHz, so we may feel secure that the "grass" about the square-wave transitions has, indeed, been caused by truncation of significant data elements in our,

TRANSFORMERLESS MIKE PRE-AMPLIFIER OPTIMIZATION — *continued . . .*

smaller samples, due to the limitation of the transfer function to frequencies below 256 kHz. Referral to Figure E will confirm that our modified beauty has significant amplitude response well beyond 256 kHz. If we wanted to verify this more fully, we might generate a transfer function over a, say, 100 microsecond time window, and plot the results of the 10 kHz square-wave response. But we won't.

For COMTRAN, as for any fine instrument, the results of its application are dependent in large measure on the skill, perception, and judgement of its user. After all, a Stradivarious is not entirely responsible for the music it produces.

Figure A: Voltage gain and Absolute Phase response for stock mike pre-amp.

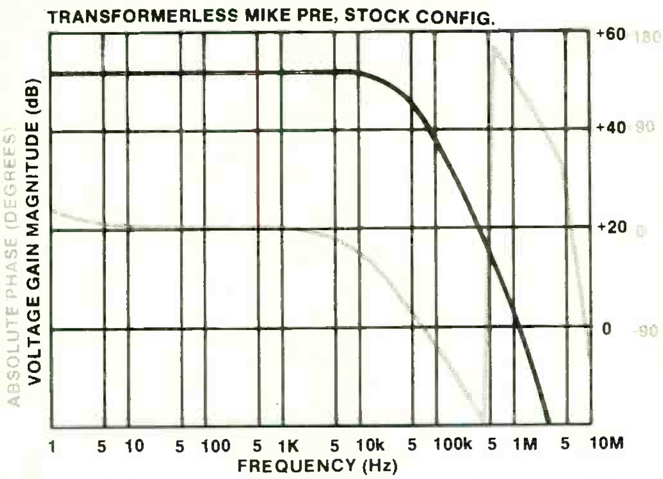


Figure B: Voltage gain and Absolute Phase response for half-differential input pair.

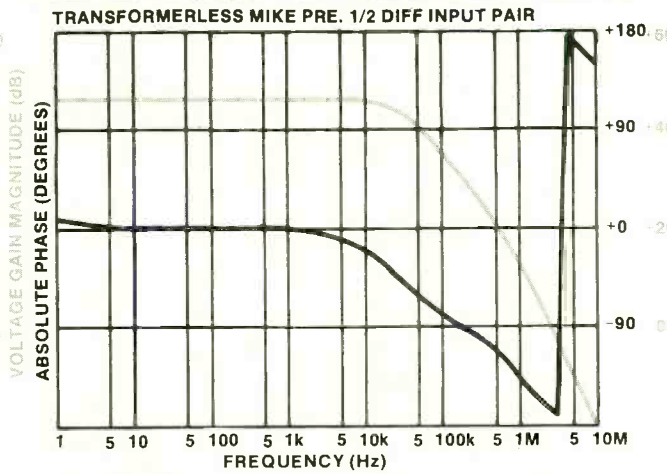


Figure C: Voltage gain and absoltue phase of mike pre-amp with phase-lead caps removed.

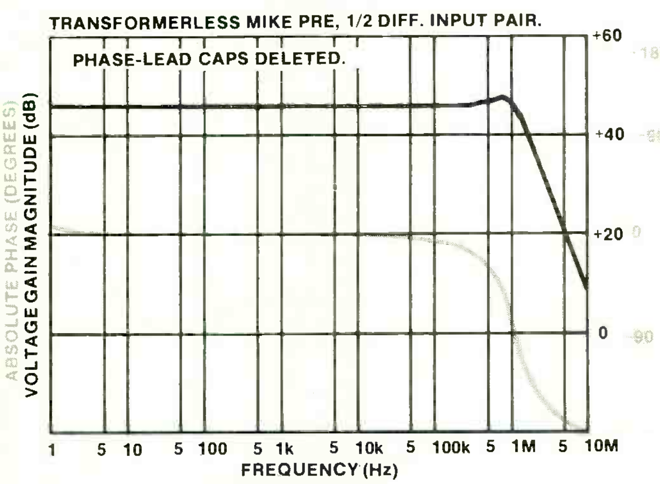


Figure D: Voltage gain and absolute phase of modified stock pre-amp.

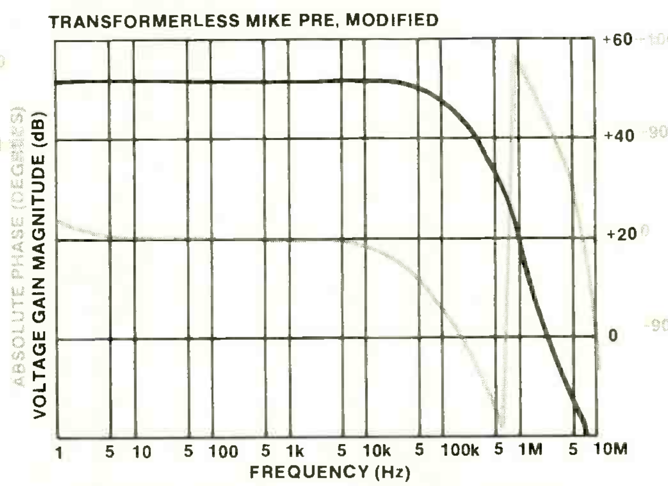


Figure E: Voltage gain and absolute phase of modified half-differential input pair.

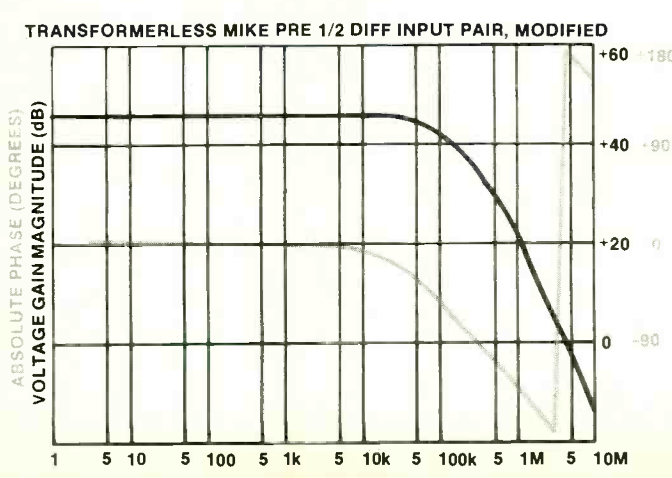
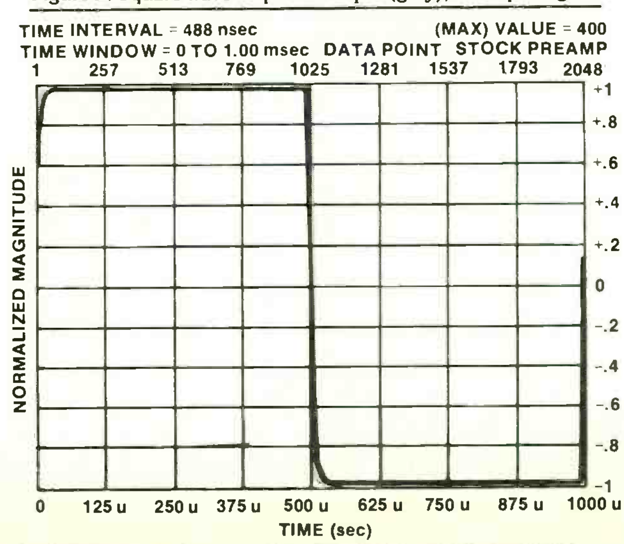


Figure F: Square-wave response output (grey), and input signal.



The more revealing Aqua Regia Cymbal test shows an even more marked difference in performance between the stock circuit and our modified prodigy. The Virginal Cymbal Crash waveform, Figure G, has been multiplied by a factor of 390.84 (51.84 dB) to show it as if it had passed through a perfect system that happens to have that gain at 1 kHz, as do our model circuits. Figure H shows the response of the stock pre-amp and in its modified form, overlaid on a vertical scale such that the perfectly amplified cymbal crash would just touch the top line of the graph at its maximum peak, as it does in Figure G. Since both model circuits have precisely the same gain at 1 kHz, we can see the differences that are obvious about the respective maxima and minima of the plots.

The differences are not merely academic. A/B listening tests confirmed that the change of these capacitor values achieved a drastic improvement in subjective sonic signal quality that altered the total performance of the console in question. Not half bad for about $2 a channel. Plus the consulting fee, of course.

References:

1. *Stabilizing Op-Amps*; Deane Jensen; *Recording Engineer-Producer*, Volume 9 #3, June 1978; pp. 42-53.
2. *Some Useful Graphical Relationships*; Richard Heiser; *Journal of the Audio Engineering Society*, Volume 23 #7, September 1975; pp. 562-564.

Figure G: Time Domain Waveform of reference cymbal crash.

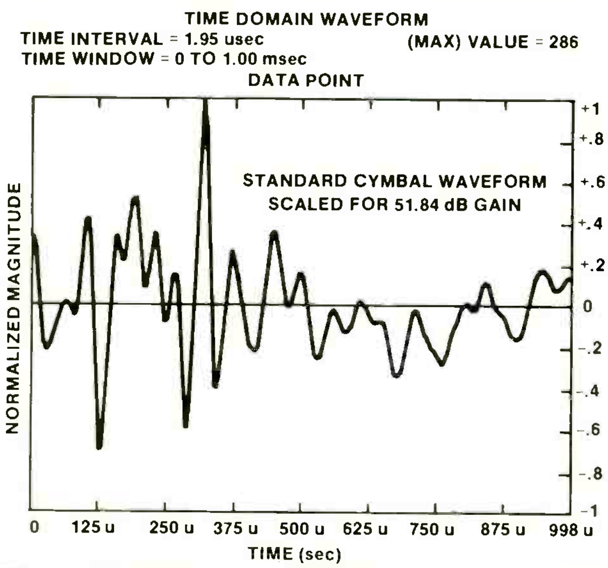
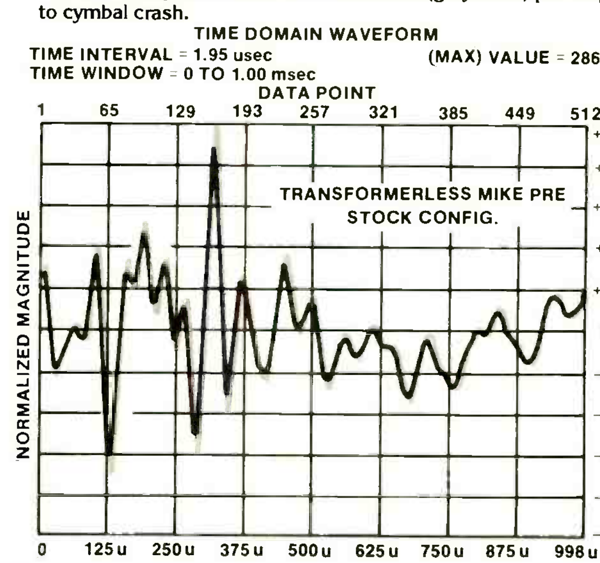


Figure H: Response of stock and modified (grey trace) pre-amp to cymbal crash.



AUDIO ON THE ROAD

MULTITRACK MOBILE RECORDING

by Gary Platt

MOBILE AUDIO: two Otari MTR-90 24-tracks, Sound Workshop 36/24 Series 24 console, and Otari MTR-12 mastering decks.

In researching exactly when the first live remote recording was attempted, you'll discover that it had nothing to do with modern electronics, or even the Twentieth Century. The true roots of mobile recording — if not all of recording itself — date back to the ancient Indians. Clay pots on hand-turned lathes were decorated by using a thin wire. While the pot rotated, a wire was held in a stationary position which imbedded into the surface not only a decorative spiral, but also sounds of livestock, and sometimes the speech of tribesmen in the near vicinity. Some clever scientists recently figured out that the pots could be "played back" in the same way using a thin wire and transducer. Although the clay surface noise was high, sounds were actually reproduced. Does that remind you of today's vinyl?

Of course, numerous live "mobile" recording dates were conducted with mono, two- and three-track formats, from the advent of the recording process itself, through the Sixties. (And, don't forget, it was not uncommon 40 and 50 years ago to come across a mobile cutting lathe, or 35mm optical recording equipment, for recording, amongst other things, film sound tracks on location.)

The majority of these latter-day mobile dates consisted of throwing some recording equipment into a rental van, and assembling it at the concert site. Many consider that the very first "permanent" mobile truck was built in 1965 by the Godfather of mobile recording, Wally Heider, in which he housed two of the very first 3M eight-tracks, linked to an eight-channel console. At the time, Heider's mobile truck was a big gamble that paid off big dividends. Every mobile unit should have a picture of Wally somewhere, so that when things go well a candle could be lit in front of him (and when things go poorly it could be a dart board).

Many would consider that the 1969 concert at Woodstock was the turning point for mobile recording —never before had such a technical nightmare been encountered. An eight-track truck was specially built to handle the concert recording, and kept the mobile crew dancing for three days and countless bands. From here mobile recording took off in a big way.

Today, state-of-the-art multitrack mobile recording means six times the tracks, eight times the console inputs, and 10 times the money of the Woodstock days.

Into The Valley of Death . . .

"Cannons to the left of me
Cannons to the right of me
Cannons in front of me
Into the Valley of Death rode the 600."

Mobile recording dates bear more than a passing resemblence to the charge of the Light Brigade. Liken, if you will, the Hollywood Bowl to the Kyber Pass, with no chance to foresee unforeseen circumstances, such as mikes falling over, torrential rainstorms, mid-concert civil disorder, or lines being kicked out. Anyone who has recorded a live concert will tell you they sometimes die a thousand deaths before it's over. All one can do is charge with the absolute confidence that their battleship is "State-of-the-Art."

So what is state-of-the-art in mobile recording? A lot of changes have happened to mobile recording in the past few years. The demand today is for a facility that can handle a multitude of stage inputs, is equipped with two 24-track machines (and maybe more), never has a grounding problem, is SMPTE capable, and provides a staff with good vibes and knowledge.

A clean, reliable console that can stand up to a few potholes and speed-bumps is the heart of any mobile system. Imagine, if you will, the anxious anticipation of firing up a console after driving it through a tropical downpour, desert heatwave, or arctic snow storm. Stability is *all* important. Console automation isn't a real necessity, because most mixing and post-production takes place in conventional, stationary facilities. "Snap-shot"-style automation, such as that developed by Harrison and Solid State Logic, is helpful for pre-setting balances during rehearsal, and having less to worry about during the gig. The *main* thing is to have plenty of inputs, bus outputs, and auxiliary outputs. Most consoles are set up with a no-nonsense signal flow that makes a quick-set situation much easier and faster. In-line monitoring helps save space in a large board, and keeps the mix engineer positioned centrally to the audio monitoring.

A matched pair of multitrack machines are a *must*, for a couple of reasons. First, unless there is some *very* coordinated control of staging, or if you're Kreskin, it's a safe bet that some part of a live date will be missed while changing reels. Second, nothing is more fickle than a multitrack tape machine. So, if one goes down, another deck is ready to roll.

Probably the one area that has seen the most improvement — and could well

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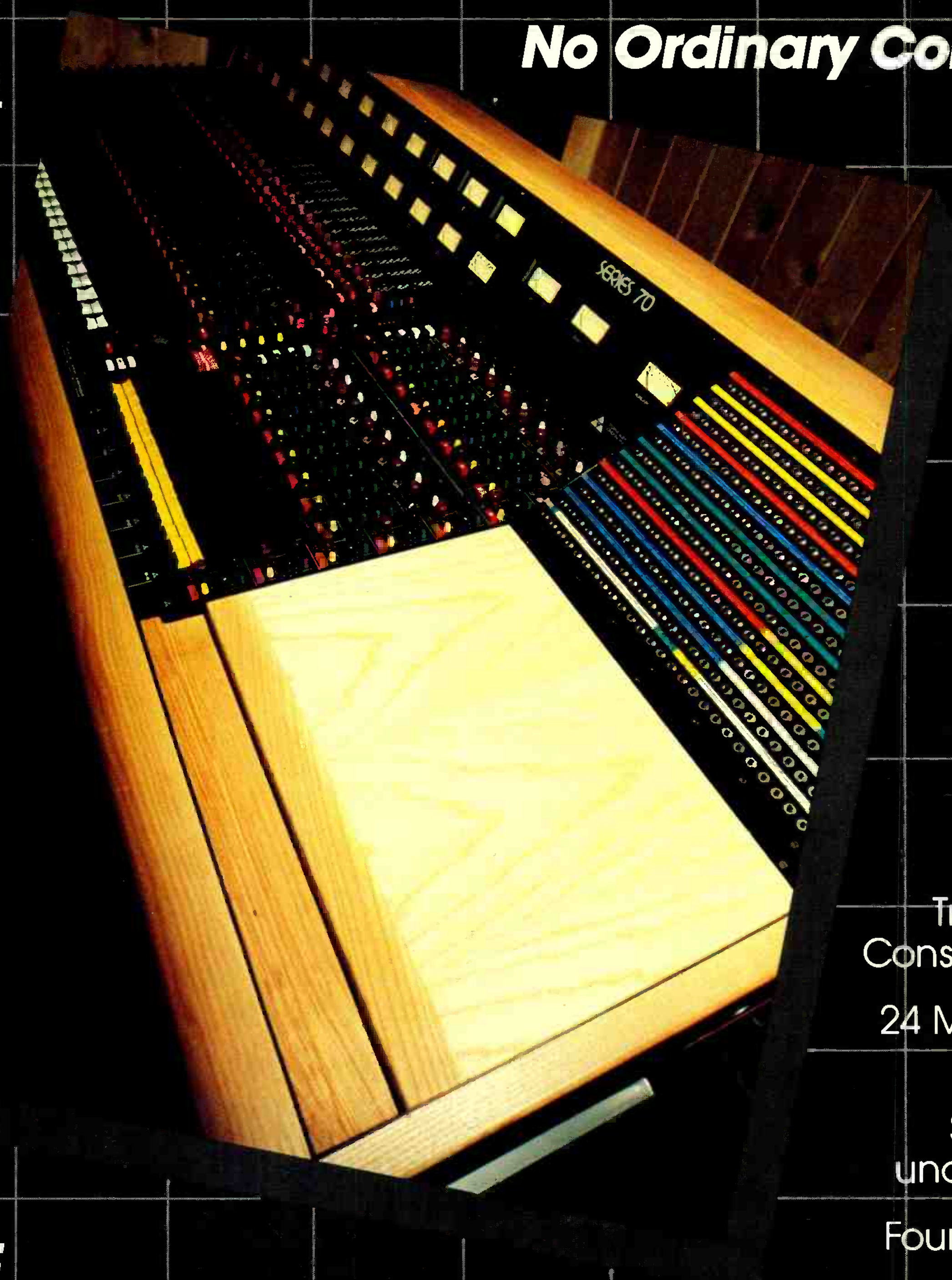

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use the most improvement in some cases — is audio monitoring. It's not easy to turn a relatively small metal box on wheels into a great-sounding control room, although a few mobiles have managed to get it right; some of the best monitors I've heard are to be found in the Le Mobile and Full Sail units. There really is *no* excuse for a poor-sounding system if it has been designed correctly. Remember, if the monitors sound like raisins, just like wine you'll never get a "vintage" recording.

As far as outboard gear is concerned, the emphasis now in mobile operations is towards providing lots of compressor/limiters, gates, and equalizers that are easily accessible from the mix position. Also, the ability to see the metering at a glance without a search is important. Units such as the dbx 900 Series, Audio + Design Scamp, and Aphex modular racks represent an excellent choice for mobile application, since a large number of devices can fit into a relatively small space. In addition, the versatility of exchanging one module for another easily accommodates different tracking situations. Mix engineers like to have plenty of outboard gear to work with, since the whole ensemble goes down to multitrack at once "en masse." The introduction of digital reverb units proved to be a Godsend for mobiles, since there is *no* way that a plate reverb could survive the traveling, and some spring units take a pounding on the road as well. Digital delays and reverb are great for adding ambience, and beefing up the live audio mix.

One of the most important items to have aboard is a number of cassette recorders. When everyone is packed up at the end of a live date, the producer, director, and musicians all need cassette copies of the live monitor mix to help in post-production.

FEDCO AUDIO MOBILE — Equipped with Trident Series 80 console and MCI JH-114 Multitracks

Fedco Audio Labs, based in Providence, Rhode Island, claims to have been engineering remote audio longer than any other studio in North America. Since 1968, the team that created the first location recording studio on the East Coast has been heavily involved with the development of both the science and the art of live audio recording. From the first major album project 15 years ago, The Doors' *Absolutely Live*, to its present workload of record, radio, television, video and film productions, Fedco is considered by many to be a pioneer and innovator in the field.

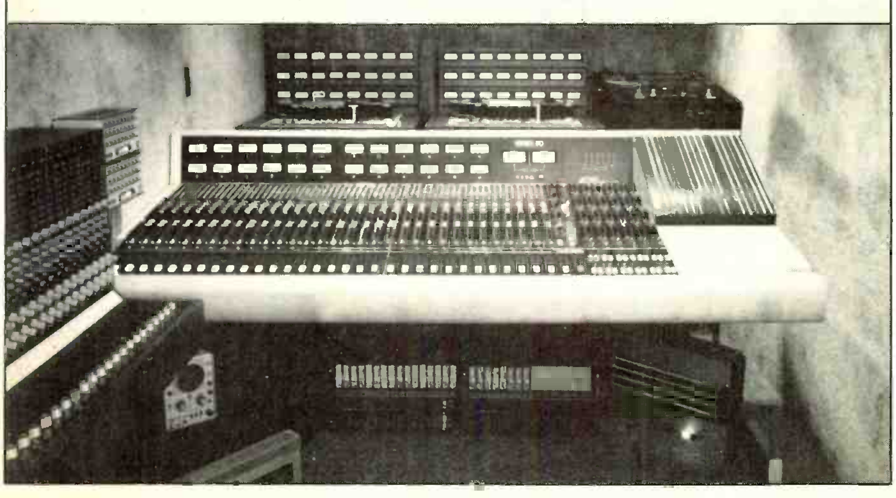
Locomotive force for the Fedco facility is a 30-foot, turbo-charged Mack diesel with a practically unlimited cruising range. The control room layout houses a Trident 32-input Series 80 console, as well as an auxiliary console designed by Fedco engineers to provide an additional 24 input channels. Two MCI JH-114 24-track machines (plus 16-track headblocks), 24 channels of Dolby noise reduction, a pair of Otari two-tracks, and Tannoy and JBL monitors complete the system.

Fedco's founder, Lyle Fain, was an enterprising physics student who often asked to borrow the family station wagon; not for cruising on weekends, but rather to put the tailgate down, make the most of space and opportunity, and record local musical groups that wanted an aural record of whatever history was being made. Progressing from two- to four-track with a 3M machine and a home-brewed console, and then up to an 18-foot van complete with custom-designed power distribution, acoustic treatment, and climate control, Fain and company helped lay the groundwork for remote recording as we know it today.

Soon followed a custom API console with a monitor section specially designed for live recording, which allowed the engineer to solo tracks one at a time or additively, and then restore normal monitoring status on command. Stevens multitrack machines then were added to complete the 16-track transition. This middle period saw Fedco's engineers involved with everything from The Mothers' *Live at the Fillmore East*, to Woody Allen's *Sleeper*, ABC-TV's *In Concert* series, Joe Cocker's *Mad Dogs and Englishmen*, and The Band's legendary *Rock of Ages* album.

Which brings us to 1975, and the initial design and construction of the present 46-track control room. The subsequent addition of MCI multitracks and Dolby noise reduction paralleled the upgrading of truck communication and closed-circuit video systems, as well as power and audio distribution. The API console was expanded to handle the increasing variety and sophistication of remote projects, and it was at this time that the company redesigned its mike-split system using Jensen transformers.

Tom Arrison, chief engineer at Fedco since 1977, explains the critical role that personnel



Power and Grounding

An entire article could be written on the different methods and opinions of obtaining power to feed a mobile truck. Most mobiles are set up with a two- or three-wire system that includes two "hot" leads for 220-volt/single-phase, and often a lead for separate system ground. In these systems both hot leads are sent from a 220-volt power source to a large transformer that delegates 110-volt current to various loads, along with a neutral lead.

A separate ground is important in the event that an item of equipment in the mobile, such as a tape machine, compressor, or virtually anything, developing a fault and sending its "hot" to the chassis of the unit. Without a separate service ground the possibility exists, although slim, that a person entering a mobile may become the actual ground path as he touches the shell of the unit while, for instance, one foot is on wet pavement.

Gary Hedden, designer and builder of several mobile units, and technician Marty Sargent, employ a four-wire, 220-volt/single-phase system consisting of two hot leads, a neutral lead, and a ground lead. One hot lead and neutral are delegated to the audio of the system, while the other hot lead and neutral are delegated to run air conditioners and the like. The neutral lead forces 110 volts across each leg of the service to run everything in the mobile. Hedden feels the advantage to be gained by such a configuration is that the entire system can be tied into any single-phase, 110-volt current, thereby eliminating the need for a 220-volt supply. Hence, any typical residential home can be tapped for power, whereas a two- or three-wire system might have a hard time, since a 220-volt service is required to run the system. It also eliminates the need for a transformer — with all its attendant magnetic field problems — and which usually weighs quite a bit.

Advanced circuit protection, such as G-MOVs (metal-oxide veristors), are installed to keep potential power transient from passing through the power system across all power leads.

Potential ground faults need to be checked in case an instrument amplifier tied to a DI box developes a ground fault, which might be sending "hot" current to a mike line on stage. A diligent crew should check with a VOM, *just* to be sure.

Some mobiles actually carry their own generator to supply power in the event of a local shortage; Le Mobile and Mountain Mobile are two examples of trucks that feature power plant generators. Such units are very heavy items to carry on-board, but they do offer the advantage that a truck then is able to record literally anywhere.

House PA, Monitor, and Mobile Interface

As will be readily appreciated, all on-stage mikes and line-level sends are sent first to a splitter box, which provides the house PA, house monitors, and mobile recording unit with individual, separated feeds to each location. In this way one basic mike setup can be used by all parties with different intentions. Unfortunately, there often can be a squabble over who gets the first split, since this feed usually presents fewer grounding problems. Quite often the recording mobile gets first choice if the stage PA hasn't already been set up.

In some cases, house PA systems don't have transformer-isolated splitter boxes, which omission can create immense problems of grounding, RF supression, and amplifier variations. To run both systems totally free from one another, transformer isolation on each mike input feed to house PA, house monitors, and recording unit *must* be installed. Usually each mike input on the splitter also has its own individual ground lift, to provide greater grounding versatility.

Sound checks with the complete lighting system, PA, and monitors fully powered is essential. Variable conduction angle switching used in most state-of-the-art lighting systems can create some awesome buzzing in audio lines.

Of course, every recording engineer has his or her own miking technique, but care must be taken in advance to make sure stands and mikes are setup in positions that are unlikely to be altered inadvertantly during the corse of a concert. Due to their wide pickup pattern, pressure-zone mikes are especially well suited for picking up audience response. If conventional mikes can be hung high overhead of the audience, and the outputs mixed with Crown PZMs placed just at the lower front of the stage, a blend that sounds particularly rich and present can be achieved easily.

The next problem often encountered during a remote session is how to accommodate the PA mixer's choice of mikes with that of the recording engineer. If a compromise isn't reached, a separate mike or two may be set up, and connected only to the recording mobile. Usually a friendly discussion over a Jack Daniels solves most differences.

Audio for Video Shoots

Current video shoots often require audio synchronization for some quick post-production "fixes" when a concert is over, for which a track of SMPTE timecode is needed. Most units carry a SMPTE generator when handling video shoots, although most often SMPTE code and 59.94 Hz sync/resolve tone is fed via separate lines to the audio truck by the video facility. If no post-concert fixes are planned, the unit need not carry a SMPTE code generator, although it might behove the more deligent engineer to take a timecode feed and print it on a spare track — just in case!

Practically all mobiles carry video monitors so that the mixing crew can see what is going on during a concert, and for improved communications with the stage crew. In the early days of mobile recording, an intercom was used between the stage and mix engineers. To keep bleed to a minimum, the mix engineer would open up the vocal mikes when directed by the on-stage engineer. Unfortunately, the time lag often could prove fatal, and the mix engineer would miss an opening line or two. A better way had to be found.

Wally Heider probably was the first engineer to use closed-circuit TV to monitor what was happening on stage. But as Heider put it, "Every time someone in the audience flashed a camera it would burn a spot in our video tubes. It got to the point where we were buying new video tubes every four or five gigs." Fortunately, the problem now has been solved with modern video cameras.

If the mobile is working on a video shoot, then the mix engineer will receive a video program feed of the concert as it is directed in the associated video facility; otherwise, the mixer views the concert from his own stationary camera set up by the mobile crew. Some mobiles

FEDCO AUDIO MOBILE . . . *continued* —

and a commitment to live audio play in maintaining high standards. "Clients come back to us," he considers, "because we're specialists in remote work. By not being tied in to one studio's way of doing things, our staff has developed the methods that succeed in a wide variety of situations."

And, as Bill Straus, remote crew chief, points out, "Our commitment to provide the highest quality service possible has taught us that when you're on the road and have a matter of hours to make a show happen, flexibility and attitude become at least as important as what brand name is on your gear. There will *never* be a substitute for giving a project 100% from start to finish."

The present truck system, though significantly larger and more complex than its predecessor, is said to reflect this commitment to quality remote work, as does the list of recording credits. The first recordings with the new truck were Gato Barbieri *Live in New York City*, and *Frampton Comes Alive!*, perhaps *the* album most recognized as putting live records on the map.

Fedco has covered tens of thousands of miles recording countless artists in sometimes far from ideal situations. Drastic weather conditions can test the limitations of any remote system, and the mobile's team has braved 108-degree temperatures in New York City to record Sha Na Na, as well as -48 degrees in Montreal to record Springsteen, and a muddy Vermont mountainside for a George Benson project. Other stand-out assignments include the Harmonium album, *L'Heptade*, at a farmhouse in Quebec; a Lear jet takeoff in quad at the Westchester County Airport, New York; sessions with Elton John, Beverly Sills, Little Feat, Joni Mitchell, Paul Simon, Bob Dylan and The Rolling Thunder Revue, Hall and Oates, Chick Corea, and The Eagles; two Rolling Stones tours; and many *King Biscuit Flower Hours*.

In the summer of 1980, Fedco used a 3M Digital Mastering System to record The Paul Winter Consort album, *Callings*. A Sony PCM-1600 digital system also was used for the Miles Davis album, *We Want Miles*, recorded simultaneously with analog in June, 1981. The Tony Awards and Miss Universe Pageant are among numerous television specials Fedco has engineered for all three of the commercial networks, as well as PBS.

In 1982, Fedco bid a fond farewell to the custom API console that had served so well for so many years. The new Trident Series 80 board is said to be extremely flexible and, in conjunction with the custom-designed auxiliary mixer, capable of handling 56 simultaneous inputs, each with full EQ. Careful control-room design has resulted in a very workable acoustic environment, Fedco engineers claim, eliminating the need for corrective equalization in the monitor system electronics, while simultaneously positioning all the recording equipment in front of the engineer within easy eyeshot.

Recent months have seen Fedco on the road recording shows for the Westwood One Radio Series, including Chaka Kahn and Joe Jackson. Other projects have included audio for Lorimar Productions' video taping of *Pajama Tops* in Toronto; The Kool Jazz Festival at Carnegie Hall for National Public Radio; an upcoming live album on the Windham Hill label; and film sound for an AC/DC project.

As the available technology constantly expands, so does the industry's demand for new ways to apply it. In the 15+ years that Fedco has been working in exclusively remote recording, it has certainly done its part to implement those advances in the field. But technology and innovation don't tell the whole story. Quality and success in location work has always required the integration of technology and the people using it into one finely-tuned system. **37 Clarendon Avenue, Providence, RI 02906. (401) 272-3157.** □□□

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- Frequency response may be altered both, at input and output of reverb section.
- Built-in limiter prevents reverb channel overloading.
- Input and output levels adjustable on the connector panel.
- Complete with remote control unit for reverb time setting.
- Switchable to mono drive of both channels.
- Isolation from structure-borne vibrations has been improved.
- Remote mix control between pure reverb signal and reverb plus individual reflections.
- Individually adjustable level for each of the discrete reflections.
- The discrete reflections are available both, as reflections only or in connection and mixed with the reverb content.
- Initial delay for the reverb signal available.
- The two discrete reflections for each channel may be adjusted in 6 ms steps from 6 ms to 60 ms.
- Available with or without Digital Control Section (BX 25 ED or BX 25 E).
- Every BX 25 E may be easily adapted with a Digital Control Section M 250.

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WESTWOOD ONE: two Ampex MM1200 24-track and ATR102 2-track tape machines; MCI JH-636 Series console with 36 inputs, and Sphere 1604 sub-mixing system; monitor system is two Altec 604Es in custom DeMedio cabinets with Mastering Lab crossovers. **9540 Washington Blvd., Culver City, CA 90230. (213) 204-5000.**

RECORD PLANT/Los Angeles: Mobile Unit #3 features 3M M79 24-track with Dolby, M64 2-track and Ampex 440C 4-track and ATR102 2-track tape machines; API 44×32 console; John Meyers monitor system and Auratone, Yamaha NS10 or JBL 4311 auxiliary monitoring. **8456 W. 3rd Street, Los Angeles, CA 90048. (213) 653-0240.**

MOUNTAIN MOBILE RECORDING: Otari MTR-90 24-track and 3M M79 2-track tape machines; Neotek Series II 28×24 and modified Soundcraft Series I 20×20×5 consoles; UREI 811-A Time Align monitor system with Electro Voice, Auratone, and Galaxy auxiliary monitoring. SMPTE timecode available. **Rt. 1 Box 25, Tulelake, CA 96134. (916) 667-5508.**

TIM PINCH RECORDING: AMEK 44 input 28/24 customized console; Ampex MM1200 24-track and AG440C 4- and 2-track tape machines; monitors are Altec 604-E with Mastering Lab crossovers and Auratones; video equipped. **6600 San Fernando Road, Glendale, CA 91201. (213) 507-9537.**

RECORD PLANT/NY: two Ampex MM1200 24-track and Ampex ATR100 2- or 4-track tape machines; custom API 44×44×24 console with built-in API 560 limiters; monitors by Westlake, modified and powered by Bryston 4B amplifiers; UREI and Teletronics limiters on truck, and other equipment supplied as needed. **321 W 44th St., NY, NY 10036. (212) 581-6505**

FULL SAIL RECORDERS: Otari MTR-90 24-track and MTR-10 2-track, and JVC DAS 90 digital processor; Sphere Eclipse custom 32-in/24-out console; monitors are Fostex LS2B and Auratones; signal processing by UREI, dbx, Lexicon, Aphex, Deltalab and MICMIX. **660 Douglas Avenue, Altamonte Springs, FL 32701. (305) 788-2450.**

GBH PRODUCTIONS: two Otari MTR-90 24-tracks, two Ampex ATR102 2-tracks, and Ampex ATR104 4-track tape machines; custom API 40×24×2 console; monitors by UREI and Auratone; Adams-Smith tape synchronizer and BTX 4200 timecode reader/regenerator. **125 Western Avenue, Boston, MA 02134. (617) 492-9273.**

LE STUDIO MOBILE: Otari MTR-90 24-track with dbx noise reduction, Scully 280 8- 4- and 2-tracks, and 3M79 4-track tape machines; Aengus 32-in/16-bus console with 24-track monitoring; monitors by JBL, Auratone, and UREI; outboard equipment by URSA Major, UREI, Lexicon, Furman and Teletronics. **715 Boylston St. Boston, MA 02116. (617) 267-2825.**

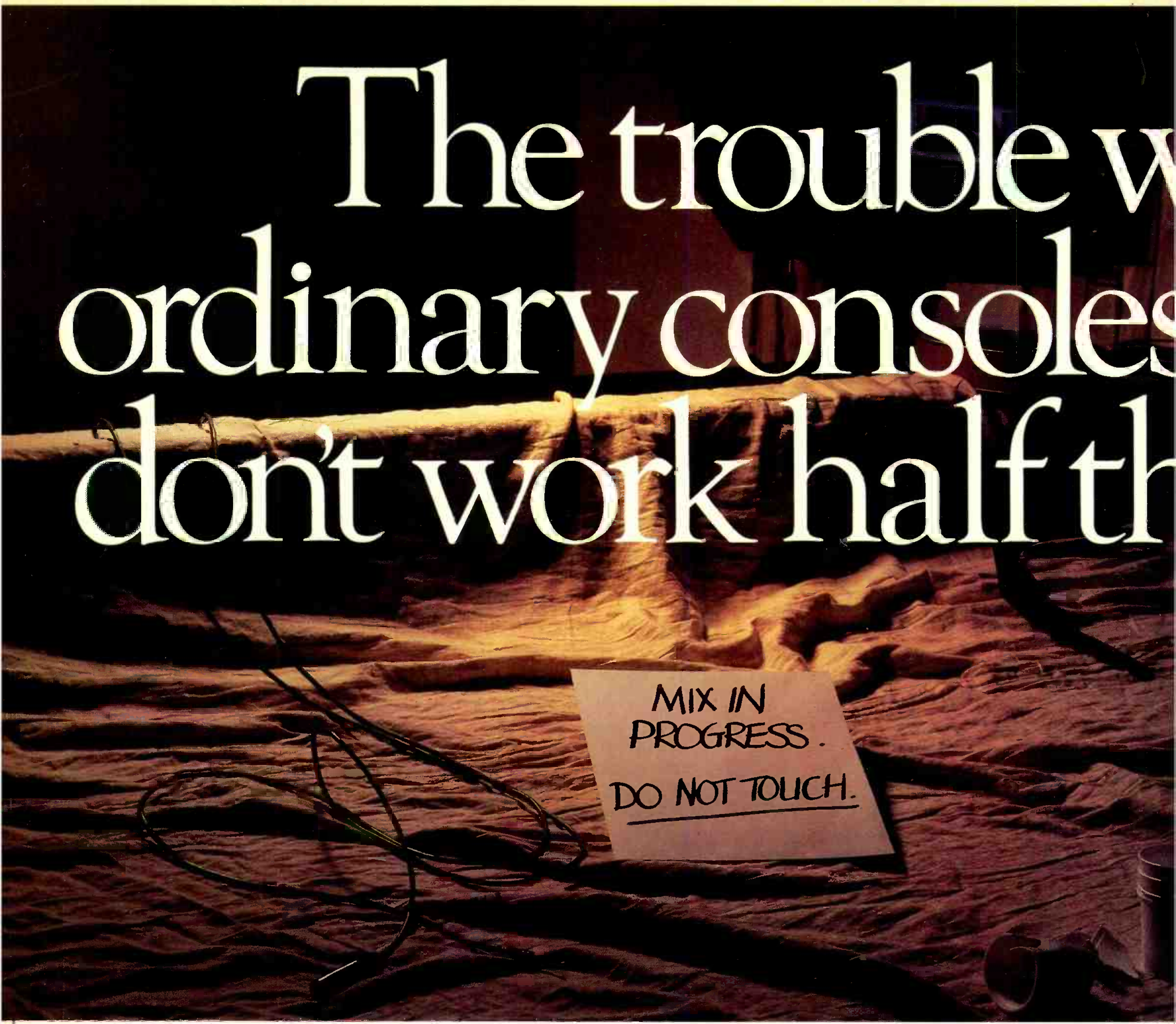
even carry two video monitors set up between the audio monitors, to enable viewing of a program feed, and its own stationary camera. In some cases, even the preview of the director's next program shot can be fed to the video monitor, so the audio mixer knows what to accent next. Also, quite often an audio monitor will be positioned next to the mixing position, so that the engineer can listen for the director's camera selections.

The Importance of a Good Crew

The most important ingredient of a successful mobile unit is a competent staff. Finding knowledgeable, stable, experienced people to work the kind of hours a mobile crew encounters is probably the hardest part of building a state-of-the-art mobile business. A great crew can be inspirational; everyone doing their job as a team. A good maintenance man (or woman) can change your religion when it's seconds before a concert, and a last-minute problem is easily fixed. Most units send a maintenance engineer, an audio engineer, and an assistant even at the extra cost. Due to the travel involved, and situations that

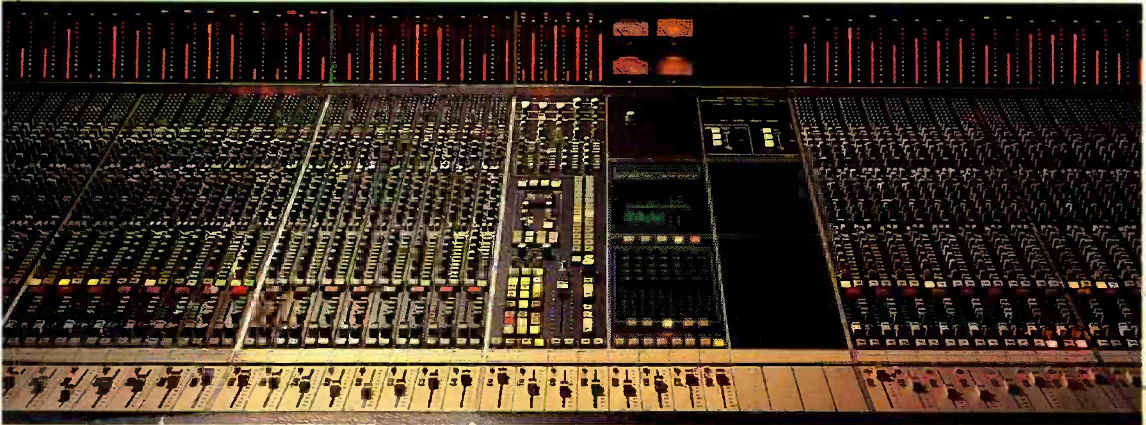
. . . continued on page 85 —

The trouble w
ordinary consoles
don't work half th



It's a situation that every studio manager recognises. A client has been in, done some work, and departed to return some time later. Expecting to find the desk as it was left.

Of course, the engineer could always note down all the settings and then reset the desk. But that's extremely time consuming and not entirely reliable.



So, usually, the studio has to stand idle between sessions. Keeping the customer happy, but not keeping the money coming in.

At Solid State Logic, however, we've developed a rather more practical solution to this dilemma. We call it the Total Recall System.

Total Recall is completely independent of all audio paths and allows the console settings to be permanently stored on floppy discs within a few seconds.

So, at the next session it takes only minutes to reload this information, check it on the colour video monitor and return the console to its original settings.

The same thing can be done at the end of each mix

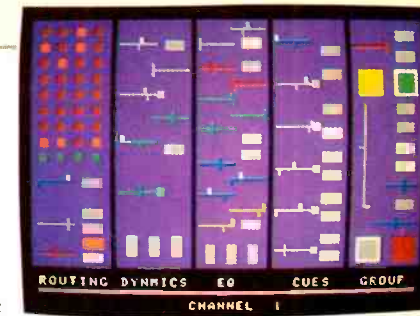
to save time at a later re-mix. And engineers can even store their personal EQ and dynamics settings and create their own libraries on floppy disc.

Total Recall is just one of the functions of the SL 4000 E's on-board computer. The computer will record all the details of a session – title entries, track lists, cue points, dynamic mixes, synchroniser information and so on – and store them on a floppy disc.

These unique facilities give the Solid State Logic Master Studio System several important advantages.

It allows the manager to keep his studio working, and earning, for the full 24 hours a day. Because even the most complex set-up can be precisely reproduced in about 20 minutes before the start of a session.

It saves the engineer wasting precious time and lets him concentrate on the creative process, from track laying to over-dubbing through to mixing. Because the studio computer speeds up everyday tasks like autolocation, drop-ins, mixing and synchronisation.



And it gives producers and musicians real flexibility and continuity. After recording in an SSL studio, they can return there (or to any other computerised SSL studio in the world) and continue work with absolute accuracy and the minimum of fuss.

Yet the computer is simplicity itself to operate. Even inexperienced assistants and tape-operators will soon master its basic functions. While feed-back from studios with SSL systems shows that more advanced expertise is acquired quickly and naturally with use.

The SL 4000 E Series Master Studio System could only have been developed through an understanding of the needs and problems of people who spend their lives in studios.

So it's not surprising that SSL's design team include not only computer and electronics experts, but engineers, producers and musicians. People who both improve studio technology and use it.

What this group sets out to devise, and SSL sets out to produce, are real answers to real problems. Finding ways to improve quality and streamline audio production. But we also produce machines that are built to last.

You will never outgrow an SL 4000 E System because you can start with the basic mainframe and then add extra modules and facilities as your needs and budget dictate. By designing systems with the future in mind we make sure that hardware and software developments can always be integrated into existing systems.

We can build you an SL 4000 E Series Master Studio System in around three months. So if you would like to start cutting the amount of time and money your studio wastes, cut the coupon or call Antony David in the UK, Doug Dickey or Piers Plaskitt in the USA.

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THE MISSISSIPPI DELTA BLUES FESTIVAL

Operating a Portable and Easily Erected Sound System for Smaller Venues

The Sixth Mississippi Delta Blues Festival, held in late September 1983 at Freedom Village, Mississippi, is an annual event representing a revival of one of the two most famous Mississippi products — Cotton and Blues. The Festival illustrates and honors an art form that was born in the delta cottonfields, and birthed by the workers and a way of life whose hardships we must never forget.

This year's Festival was sponsored by Mississippi Action for Community Education, Mississippi Division of Tourism, Winthrop Rockefeller Foundation, The Ford Foundation, the Mississippi Arts Commission, the Miller Brewing Company, and Peavey Electronics Corporation. Featured artists included blues masters John Lee Hooker, Z.Z. Hill, Larry Davis and Bobby Rush, along with Eugene Powell, Son Thomas, the Nighthawks, Lonnie Pitchford, Sam Myers, Lynn White, Sam Brothers 5, and Lefty Dizz and Band with "Queen" Sylvia Embry.

According to Peavey sound crew member Robby Reece, the company was chosen to provide sound reinforcement equipment for this year's Blues Festival because "Hartley Peavey feels that all forms of 'American Music' had its start in Mississippi. The 'roots' of today's contemporary music is the Blues. Hartley Peavey has lived in Mississippi all his life and built his electronics company from a basement operation into a major 'Made in USA' American manufacturer — all in Mississippi. He chose to sponsor this year's Festival and provide all the necessary sound equipment as his way of saying, 'thank you, Mississippi and especially the Blues, which started it all.' "

Sound System Layout

The main speaker placement was of particular note because of its lack of symmetry — there was one loudspeaker array rising some 20 feet from the ground to the right side of the stage. Pressed for an explanation of the single-stack approach, Peavey's Marty McCann says that "this venue is one large rectangular former cotton field, and as you face the stage it is located in the right-hand corner. Therefore, 180-degree or even 120-degree dispersion is not necessary.

"For years every sound company has been erecting stacks and stacks of speakers on each side of the stage because that's the way it has been done, I suppose. However, this is an incorrect approach resulting in phase-related ragged response generated by the considerable distances separating the two stacks. This 'comb filter' effect has been discussed frequently in the literature from as long ago as the Fifties. It is accepted that the ideal placement would be an array flown directly above the stage, which is not always possible, particularly outdoors where there is no roof from which to suspend the loudspeaker array, and especially when the budget won't allow for scaffolding and rigging.

"Since the Delta Blues Festival stage is located in one corner, it actually presents a nearly ideal situation for presenting a model of a 'single-source,' three-way focused sound system. If this had been a venue with the stage in the normal central position, this approach would have been a little more difficult, and would have required substantially more sound equipment. I guesss the theme of this approach would be 'Less is More.' "

From a number of vantage points within audience view of the stage, many Festival attendees considered that the single-source, three-way array was indeed offering complete and totally adequate sound coverage.

Loudspeaker System

The main system was designed to operate as a three-way single source, and consisted of components that make up the Peavey Project One Series. System low-end was made up of six FH-2 low-frequency folded horns, stacked straight up from the ground in a vertical array measuring nearly 15 feet high. Each FH-2 features two Model 1504 Black Widow/Super Structure loudspeakers capable of handling 300 watts of continuous power. Each FH-2 has a quoted bandwidth of 60 to 400 Hz, ±3 dB. In this configuration, the 3 dB down point on the low end of the system was 45 Hz, due to the mutual coupling of the horns creating an equivalent larger folded horn with an increased transmission coefficient.

System midrange consisted of three MB-1 mid-bass horns stacked vertically on a 10-foot scaffold next to the FH-2 array. The MB-1 mid-bass horn features a special design Model 1202 Black Widow/Super Structure loudspeaker, and is said to offer radiation geometry of 60 degrees in the horizontal plane, and 30 degrees in the vertical plane. The bottom MB-1 was propped up from the rear and thus angled downward, the middle and top MB-1 being shimmed up from the front in order to reduce overlapping of their vertical angles of coverage and minimize the lobes that would otherwise be introduced in the horizontal pattern of coverage. Each MB-1 has a quoted bandwidth of 150 Hz to 1.2 kHz, ±3 dB, and is capable of handling 250 watts of continuous power.

Three CH-4C high-frequency horns completed the high end of the three-way system. The CH-4C horn has a constant geometry of 60 degrees horizontal by 30 degrees vertical angle of coverage that is said to be uniform with frequency response, and provides a consistently optimum pattern control within its rated frequency range of 800 Hz to 16 kHz. The high-frequency driver is a Model 22-A compression driver rated at 40 watts continuous pink noise. The three CH-4C horns were stacked vertically on top of the FH-2 array. The bottom horn was angled downward from the rear, and the middle and top CH-4C shimmed up from the front to reduce overlapping of their vertical angles, and ensure a smoother, lobe-free horizontal angle of coverage that proved to be uni-

Alabama members Randy Owen, Teddy Gentry, Mark Herndon and Jeff Cook accept the Ampex Golden Reel Award for "The Closer You Get." The record also won "Album of the Year" at the 17th Annual Country Music Association Awards show.

have to capture that with either a limiter or gain riding. Since I use very little limiting — next to nothing on his vocals — I do a lot of gain riding. But after I've heard his part a couple of times, I know where the high and low levels of the performance are.

"Randy is also an unusual singer, in the sense that he can sing for very long periods of time, like 10 a.m. to 8 p.m., without showing any fatigue in his voice at all. Once in a while though, when it does show up, he loses the bottom, and I have to compensate by adding more EQ in that range. From time to time I've had to add maybe 2 or 3 dB around 100 to 120 Hz. I may do that, too, when I think he is a little too far from the mike, or for those songs that have him in a higher pitch, and maybe calls for a little more warmth. For the 47, I recorded him basically flat."

Setting up the Cue Mix

"The drummer is in a booth, so for Randy to feel the music, he wears 'phones. He may be only several feet from the rhythm section, but everybody here in Nashville plays pretty soft; there's not a lot of activity in terms of sound in the room.

"Generally, the vocalist always wants to hear more of himself than the band. We send a three-way cue out to the studio during tracking sessions. The band is a stereo mix, and the vocal is on a separate line, so everybody can turn the vocal up or down depending on how loud they want to hear it."

Background Vocal Techniques

"To the best of my recollection, Jeff Cook and Teddy Gentry, who sing the backgrounds, have never kept the vocals from a live performance. They always go back in the studio to rework the parts, and modify them to some extent. So the background vocals are almost always overdubbed.

"Teddy always sounds best on a Neumann U-47 for low harmonies, and on an SKM5 Studer for high harmonies. Jeff likes a Neumann U-87, and that combination makes a blend that requires very little equalization. I try to go for the mikes that capture the way I want to hear the vocals coming back. Then if I have to tweek the EQ, I do it."

Special Vocal Effects

"There really is no doubling for the lead vocal, but on *The Closer You Get*, we used just the slightest amount of Harmonizer, to give the track more of an 'electronic' sound. The guys are very conservative when it comes to effects. If they use any at all, the effects usually are added on a considerably less than 1:1 basis.

"In almost every instance, the background singers [Jeff and Teddy] are doubled. But sometimes, the combination of two Jeffs and two Teddys with one track of Randy presents a blend problem from a mixing point of view. The sound becomes somewhat disconnected. When that shows up, which is not often, I double Randy's voice electronically with a Harmonizer or a Lexicon delay set at like 25 or 40 milliseconds, just to add a little bit of doubling effect.

"On a tune called 'Dixieland Delight,' there's a breakdown chorus about three quarters of the way through the song. That acappella section has a rather drastic pitch change [1.012] on it, which would have been offensive on an instrument like a guitar or piano. But vocally, the relationship between the live sound and the electronic sound was quite pleasant. The two signals were mixed on a one-to-one basis."

Placement of Vocals in the Mix

"We really take each song as it comes. On 'Lady Down on Love,' we didn't double until the very last phrase that had harmony. In that respect, it was quite necessary to keep the voices mixed tightly together until the very last title was sung out. Then we doubled the parts and went to a stereo spread on the vocals. When I spread the vocals for Alabama, I don't split the doubled parts and put one of each side of the mix — I like a double high-vocal left center, a double low-vocal right center, or vice versa. Then the lead vocal goes dead center, and that's either a single or a synthetically doubled part."□□□

makes that part happen. For example: suppose the original interpretation was to sing intimately to make the verses sound right, but the chorus just didn't come alive that way, because it's difficult to let loose after holding back on the verses. But, on another day, the artist may sing the whole song harder. Although the verse may not be quite right, the chorus part might be *perfect*. So you cut the two together to get a spectacular version —soft verses with harder-sounding choruses. Whether the discontinuity is a disadvantage or advantage really depends on what you need to make the song work at that point in time."

Signal Processing and Outboard Effects

"For many years we've heard what an acoustic guitar sounds like. People want to hear new and different sound qualities. So the whole philosophy of the Eighties is to come up with those new voices. That's why electronic instrumentation, like the Vocoder, is becoming so popular, [because] it can give a human voice a totally electronic sound for lead parts or backgrounds. But, the electronics can give the music a feel that's too *cold*.

"Capturing a warm sound in a voice all starts with a great singer; the *singer* must deliver what you want to hear. If the vocal needs to be soft, with an intimate quality, the artist really has to sing that way. I always try to get the difference out of the singer, as opposed to arriving at the change by using outboard gear, different mikes, or equalization. It's just about impossible for the engineer to change radically those qualities of expression, and still maintain the reality of the vocal.

"What can be changed for variety and emotional effect, however, is the tonal quality among the verses, chorus and bridge. That I can do with outboard effects or EQ. I may have a short DDL from a Publison, or short decay time on the verse. A great example of this is the 'Major Tom' record by Peter Schilling. The chorus has a long delay on the vocal —maybe a 7½ IPS tape slap, or a long DDL — so the chorus has a different feel.

"We hardly ever record the vocals with effects, because we never know how we want it to sound in the final analysis. As I mentioned before, sometimes the singer needs to rerecord a line or two a couple of weeks later. Leaving the effects off the vocal track means there's less to duplicate for the second session, which is especially important if we're doing any complex composite work." □□□

ANALYSIS OF A SESSION

Talking with Gene Rice, Engineer for Alabama's *The Closer You Get* Album

"Randy Owen is the lead vocalist on most of the album cuts, and always sings lead on the singles," Rice points out. "He very rarely, if ever, overdubs vocals, because he never seems to sing nearly as well. He finds it much easier to project the emotion as a live vocal performance. When I'm recording him, I don't get a second shot. Even if his voice is a little on the ragged edge, he'll keep the take if he feels he has the performance there.

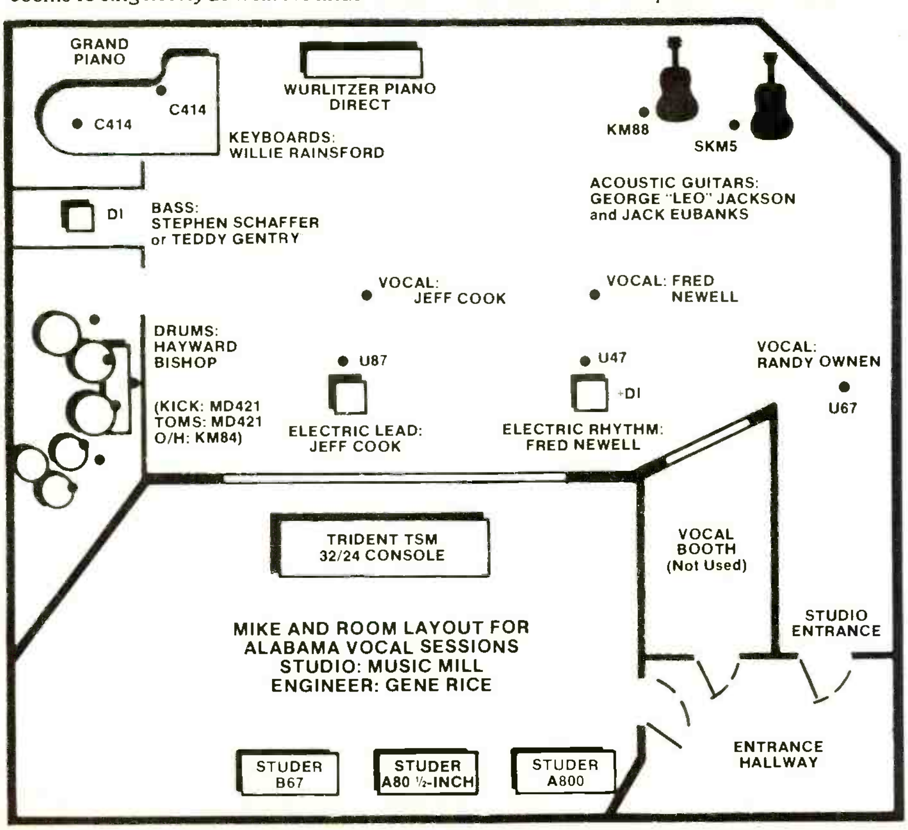
"For the last album, which was recorded at the new Music Mill in Nashville, I attempted to isolate the lead vocal in a booth, but Randy said he felt disconnected from the other players — as if he wasn't a part of what was going on in the studio. He likes to physically *feel* the music. So I had him out in the main room about 10 feet from the rhythm section.

"Up to, but not including, the last album, Randy customarily sang into a new Neumann U-47. Very early in his career, he started with the 47, felt comfortable with it, and stuck with it. I only mixed the first two albums, but with the third record [*Mountain Music*], I started cutting the tracks, too. I continued to use the U-47, but wasn't very pleased with it, because his voice and that microphone didn't seem to be the right combination that I wanted to hear.

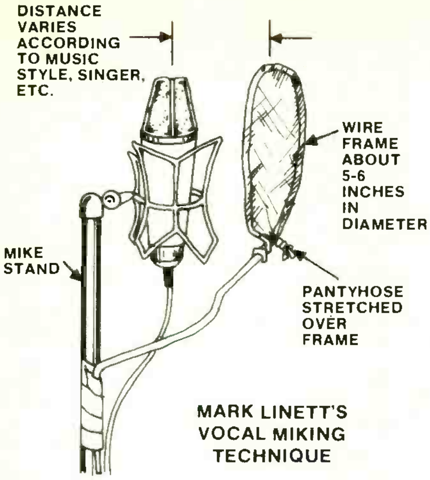
"On the last album [*The Closer You Get*] I tried the U-67, which is what I felt he should have been using all along. Randy projects quite a bit, and has a lot of upper-mid frequencies in his voice. The combination of the 47 and his voice always seemed to be on the 'edgy' side, although it didn't stop them from having Platinum albums. I just felt that the warmth of the U-67 seemed to be a very nice offset to the top-end of his voice. He still projected and had a nice edge, but there was a little bit of 'roundness' that was more in line with what I wanted to hear. We stayed with it for the entire album.

"The EQ on the 67 was not a lot different than that used on the 47. I ultimately wound up giving Randy a little dose [about 2 or 3 dB] around 8 kHz, with a very sharp peak to bring out the very upper edge, the sibilance area, of the vocal. I tried to pin-point the boost in the upper range for clarity to make the vocal project, and at the same time avoid any of the upper midrange at all, which gives you the peaky sound.

"Randy works a studio mike very well; he doesn't swallow it like a lot of vocalists tend to do. He stands eight to 10 inches from it, and that eliminates a lot of proximity effect. But if there is anything to be captured in terms of a low-volume note or a high-volume note, I



MIKE AND ROOM LAYOUT FOR ALABAMA VOCAL SESSIONS
STUDIO: MUSIC MILL
ENGINEER: GENE RICE



MARK LINETT'S VOCAL MIKING TECHNIQUE

structed of pantyhose and a coathanger may save some money and headaches.

"The coathanger is formed into any shape — a circle, square, or whatever —of about five to six inches in diameter, and the pantyhose is stretched over it [see accompanying diagram]. Then the structure is used as a sort of fence to keep the vocalist back from the mike the desired working distance. The vocalist may sing right into the pantyhose as he would with a normal windscreen placed over the microphone capsule, yet the distance may be varied to suit the recording requirements. The thin, double mesh eliminates the detrimental effects of the pops and clicks by diffusing the breath."

Kevin Clark suggests that an engineer might try varying the tape speed when the artist doesn't quite have the range to hit the highest and lowest notes of a particular performance. By slowing down the speed of the tape, the pitch drops to within the singing range of the vocalist. Once the part or parts are recorded, the tape is brought back up to speed.

What It All Means

Ironically, the mark of a talented vocal specialist is often the *absence* of a particular sound. "Great recording engineers, like great literary agents, don't leave fingerprints; the process is transparent," says George Massenburg. "Even when an engineer listens to a record, he or she shouldn't be thinking about, 'What mike was on the vocals?' or 'Man, that's a big echo!' When the engineer is good, everything becomes *invisible*. There can be a tremendous ego difficulty with that.

"Part of what [a good engineer] provides is a particular sensitivity to the intrinsic quality of sound. My aim is to keep my craft in the background by making the technical aspects of my performance invisible."

And that's the art of the recording engineer.

Recommended Reference: *The Physics of Music;* Scientific American reprints; W.H. Freeman and Company, New York City, 1978. ■■■

COMPOSITE VOCAL SESSIONS — *continued . . .*

nary performance, is happening a lot in today's music, as a result of the versatility provided by the multitrack environment. The vocalist, if he or she is an okay singer, has the option to do a vocal track more than once, and get an exceptional performance. Personally, I feel it's best to work with the *least* number of tracks. After a certain point, the process gets pretty confusing, especially if you're dealing with a great artist who has given you two completely different performances, and both are spectacular on the same line. Then you have to make a decision as to which one is best and, most importantly, which helps the record out. Working with more than five tracks gets ridiculous. I find that between three and five tracks is usually optimum.

"Most of the takes are recorded at the same session. However, we may do a composite vocal on one day, and a few days later discover that a couple of lines need to be sung a little differently, because the composite doesn't fit the instrumental interpretation of the song, or the performance may not be quite right. At that point, it's best to have the vocalist sing another full vocal pass from the top to the end, even though you're looking to replace only a couple of lines. It's not unusual to find that the artist delivers a much *better* performance after they've heard the original version. Often the artist may interpret the song one way, while the producer and engineer hear it another way. But, once the instrumentation is fairly complete, and the composite vocal track is assembled and pretty close to what the intended interpretation should be, the singer can go out there and nail it in one pass, with only a couple of punch-ins for a word or two. The composite actually may give them another new perspective on the performance that wouldn't be possible any other way.

"Now the biggest drawback to doing additional takes at a later time is reproducing all the elements of the original recording environment to match the sounds [of the subsequent takes]. After you listen to the old tracks and decide to go with that sound, the engineer has to replicate the same miking distance, [and] set up the same mike in exactly the same place with the same type of EQ, limiting ratios, etc. The matching is difficult, but that's why engineers who are vocal specialists get paid the big bucks!

"Unfortunately, there are characteristics that change from session to session that can't be controlled. The time of day may be different [morning, afternoon or evening session]; what the artist ate for breakfast or lunch; their emotional state, etc. All of these factors can affect the vocal timbre and, consequently, alter the sound of the individual tracks.

"Fortunately, that can be an advantage if the engineer and producer look at it with the right attitude. At a certain point in the song, that new sound quality may be *exactly* what

MANHATTAN TRANSFER'S *BODIES AND SOUL* — *continued . . .*

the mix to two-track. During the final music/vocal mix, I just adjusted two faders.

"The recorded tracks had very little EQ — if I remember correctly, I worked with mike placement only. I think probably 50% of the vocal recordings were completely untouched, except when the group decided to get involved in the mixing after I had completed the original mixes. They kept asking for more EQ at about 3 kHz. They said that was going to make the vocals [sound] wonderful. I ended up doing some equalizing for them that was not really something I would care to do, and some of the tracks sounded overly bright to me.

"We used outboard effects only on the pop pieces. I limited the vocals lightly [1 or 2 dB of compression] to make my job a little easier later on. When someone is singing either very loud or very soft, the dynamics are sometimes hard to control. But my limiting really wasn't doing much of anything — just evening out all the parts. I never relimited during the mix."

□□□

Danny O'Keefe has been singing professionally for more than 15 years, and writing songs almost as long. He's recorded four albums for Atlantic Records (*Good Time Charlie's Got the Blues* came from that association), and Warner Brothers. He splits his time between Hollywood, and his home on a remote island in Washington state.

artist's motivation, warns O'Keefe. "When I go into the studio, I prefer that the engineer have everything — microphone, headphones, the cue mix, etc. — ready and working, so I don't have to wait. I can just start singing and concentrate on the performance. Ideally, a lot of singers, who know their material very well, use the first two or three passes to relax themselves, and these are usually the great takes."

The Perfect Recording Arrangement

At the risk of alienating some fine producers, says George Massenburg, "the perfect recording arrangement is for the engineer and the artist to work alone, because the engineer has to deal only with a headphone mix. He doesn't have to please anyone else in the room; he can hear exactly what the artist hears, and react immediately accordingly."

And what instruments or sounds should be in the cue send? "I like to hear keyboards for pitch; drums for being able to stay in the pocket; and, of course, the bass," says vocalist Paulette McWilliams. "Sometimes the sweetening can throw you off, so I don't always like to hear that."

Danny O'Keefe agrees: "In addition to the drums and bass, I like to have the vocal track right out there, so I can hear all the increments of my own voice. But it has to be balanced with the instrument that carries the most tonal information, such as piano. I don't need the guitar, or the sax, or even the other background vocals, unless I have to blend to them. Primarily, I should have the parts that give me the most direct information that is relevant to my part."

What that perfect vocal level is varies according to the artist's preference. But there are some concrete guidelines for the engineer to follow. "Where a sound is placed levelwise in a mix changes the way the performer sings, and the type of performance you get on tape," says Massenburg. "Basically, I think that if the voice is monitored very loud in the headphones during recording, and is played back at moderate levels, almost consistently it will sound flat in the mix. Of course, a lot of insecure vocalists want their voices boosted up in the cue send, so it's important for the engineer to be listening through headphones too, while doing the recording. That allows one to find the optimum level where the vocalist can hear well, but also be able to maintain an intonation that matches the track. Often, if the singer has consistently bad intonation, a better headphone mix could improve things."

Special effects, too, are generally considered counter-productive. "I don't like to have my voice 'glossed' too much," adds O'Keefe. "I tend to hear the increment of the effect, rather than the increment of my voice. I can't be as critically aware of myself going a bit sharp or flat."

Tricks of the Trade

Some vocalists, who are used to singing close to the microphone in live situations, bring those habits to the studio, where the microphones are more sensitive. **Mark Linett**, a staff engineer at Warner Brothers' Amigo Studios, North Hollywood, describes a windscreen idea he got from Chet Himes, Christopher Cross' engineer: "A microphone may tend to overload and distort, or even be seriously damaged, by the severe transients of vocal 'pops' and other singer-induced causes. A windscreen con-

ANALYSIS OF A SESSION

Talking with Gary Skardina, Songwriter, Engineer, Studio Owner and Vocalist

"The engineer's first step toward recording a great vocal track," Skardina suggests, "is choosing a mike that's a good match for the singer. Melissa Manchester, for example, has a nice warm voice, so I use a 1950 Neumann M-49 tube microphone with a warm sound and a lot of body. The Telefunken 251 is another tube mike that I use a lot. Usually, once we've chosen a microphone for a song and everybody likes the sound, we stick with that choice. The only time we may change something equipmentwise in the recording process is if the vocalist alters their approach radically; they may go from a very intimate to an exceptionally hard interpretation.

"The type of performance determines the miking distance, too. Generally, because I like to do vocal tracks as overdubs, I can position the mike and the singer about eight to 10 inches, maybe a foot apart. Of course, if they're singing *really* hard, I back them up a little more, and it actually sounds better, because the proximity effect of the cardioid pattern doesn't get a chance to build up as much.

"If I want a lot of bass response — a real 'Neil Diamond-type' vocal — I bring them very close to the mike — within a couple of inches — to achieve a soft, intimate performance. But that closeness increases the susceptibility of picking up much more lip 'smack,' breath and vocal pops, as well as drastic changes in the amount of vocal presence if the singer moves away from the mike a little bit. In those cases, the singer bears most of the responsibility for maintaining control over the sound quality. The engineer can really only remind them to be aware of how they're working the mike."

Composite Vocal Sessions

"Doing comps, or combining the best of several vocal tracks to produce one extraordi-

Kevin Clark comes from a family of musicians, and has done a certain amount of session work around LA himself. About seven years ago, John Baylor, a successful Los Angeles singer and arranger, asked Clark if he would like to work at Baylor's home studio as an engineer. He and Baylor are now partners in Tape Recorders, Inc, a 24-track room in Hollywood. Clark is an independent engineer, but does about 50% of his projects in his own studio. Recent work includes jingles and commercials for Budweiser, Western Airlines, Continental Airlines, Levis, the 1984 Olympics, as well as record dates for Quincy Jones (some tracks for *The Wiz)*, Manhattan Transfer, and Seawind.

as a subconscious contact with another person or audience.

"So any shift or drift in intent, which is the momentary absence of a consistent, focused energy while performing, has a significant impact on the listener's subconscious, and the artist loses their attention. An actor may describe it as falling out of character for only a split second, but that's *all* it takes to break the magic. A true vocal artist can, without music, give lyrics great life by magnetizing each word, each moment, and maintaining a consistent character for the duration of the performance."

But to hold intent, or a specific feeling, for an extended period of time is difficult, and most people can't do it easily. Barigian recommends that an engineer can help in such a situation "by demanding much more expressiveness on every word. Don't let the singer start off until they have a great intent, and make every moment in each phrase significant. Once the intent is significant, don't let the singer come to rest between the words. The difference is *incredible.*"

Singer/songwriter **Danny O'Keefe** adds a personal perspective: "In a concert situation the singer has the advantage of drawing on the audience's energy for inspiration. In a studio, he has to generate that excitement completely from within himself, and it's much harder. I'm not only trying to consciously approach the song to make sure all the technical aspects of the performance are correct, but unconsciously approach it at the same time, so I can lose myself within the song and call forth the emotion that brings the song alive."

Singing can be very much like method acting, O'Keefe says, in the sense that the singer must prepare for a song like an actor prepares for a scene. "You have to be able to imagine the intended situation that the song is talking about, and charge the song with emotion. A good singer usually has some kind of psychology that they use on themself to get set for the recording, and that method varies depending on the type of date, and the personal preference of the artist."

By understanding the importance of an artist's preparation and motivation, an engineer becomes more aware of his or her own responsibilities — to protect, nurture, and capture the intangibles of a great performance accurately, and without effort.

Hanging out in the studio before a date is one condition that deflates an

After spending a few years in A&R at ABC Records and 20th Century Records, **Gary Skardina** decided to open his own studio. That was in 1975, and since then Music Grinder in Los Angeles has provided him with the opportunity to engineer and produce dates for the Pointer Sisters (he also placed one of his songs, "Jump," on the album), Melissa Manchester, Chaka Khan, Lou Rawls, Jean Luc Ponty, Donna Summer, and many more.

MANHATTAN TRANSFER'S *BODIES AND SOUL* *— continued . . .*

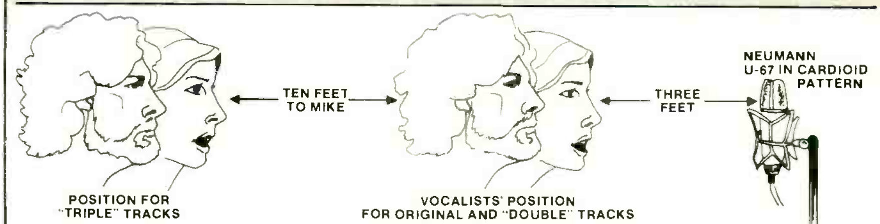
putting her right on top of a U-67, and she blew up two of them. She'd do a 'p'-pop, like 'please,' and when all that air hit the microphone, there was no more edge. The capsule either collapsed, or something in the power supply gave out. In the middle of a vocal, we'd have to change the mike. That's why I went to U-89s, which are FET-amplified mikes that can take the higher level much better. It was a very expensive lesson for me.

"On the 'Spice of Life' single, I used a Neumann U-89 about a foot away from the two girls singing lead. Actually, Cheryl has only one line just before the harmonica solo, where she sings sort of a cameo part. The backgrounds were U-67s [cardioid pattern] with the singers back about three feet, and both of them on the same side of the mike. Eventually they all sang background on the song, but it was the process of two girls singing the lead note of the harmony section on one pass; doubling it on the second pass, and then tripling the same part. For all the triples I backed the singers off the mike a good 20 feet. That way the triple added a substantial ambience to all the parts.

"Then we'd go down the line. If we had four-part harmony, we'd have 12 tracks just for that. The girls would sing, double and triple the first and second harmony parts; the guys did the same for the third and fourth harmonies, for a total of 12 tracks of background vocals. Sometimes there were five and six harmony parts."

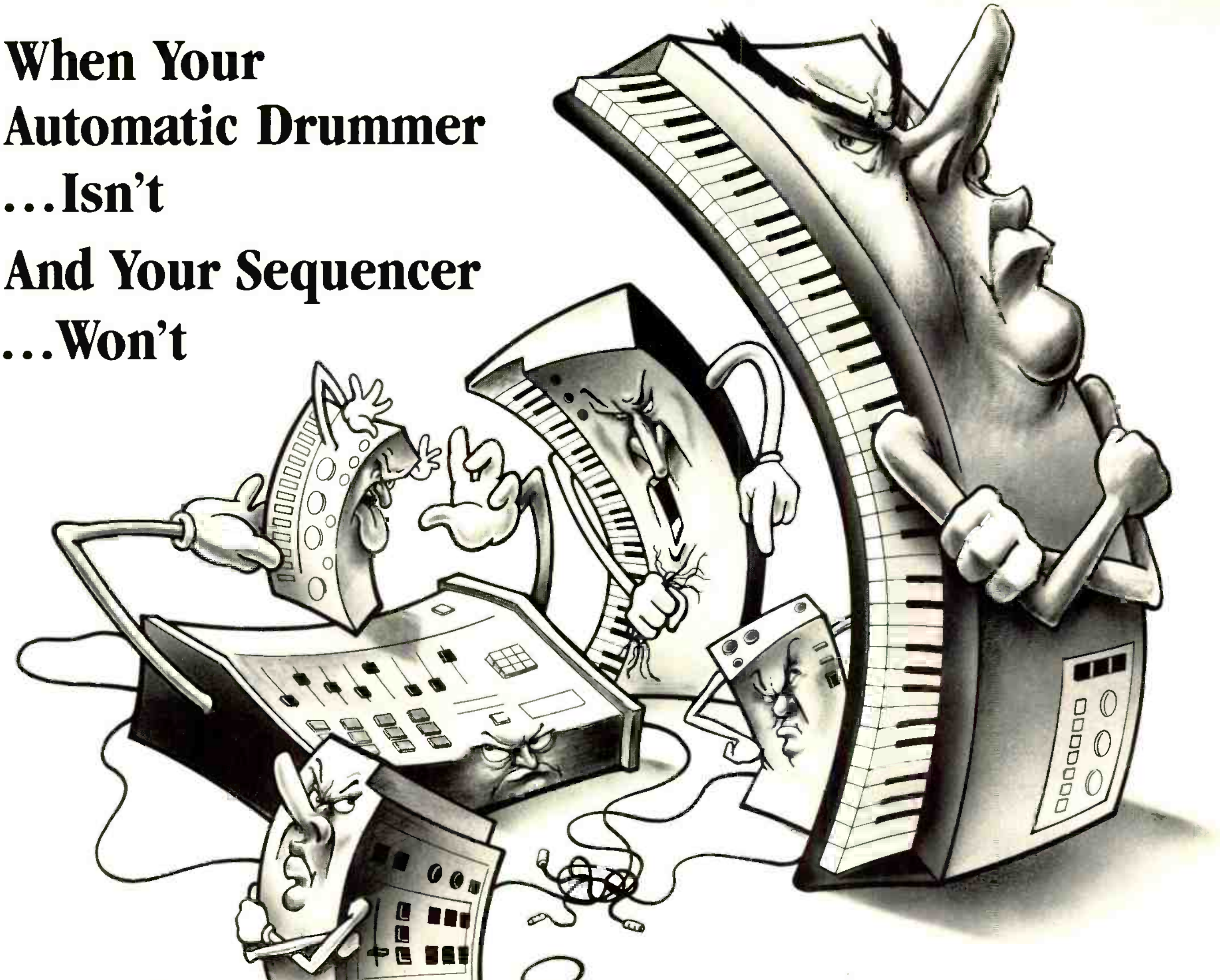
Console Automation and Signal Processing

"There wasn't a big need for automation. Rod and I bounced down all the vocals from one 24-track to the main multitrack, and I had individual control over them as I was doing



RECORDING LAYOUT FOR BACKGROUND VOCALS ON "SPICE OF LIFE," FROM *BODIES AND SOULS* ALBUM BY MANHATTAN TRANSFER

When Your Automatic Drummer ...Isn't And Your Sequencer ...Won't



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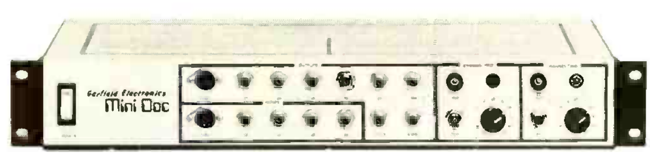

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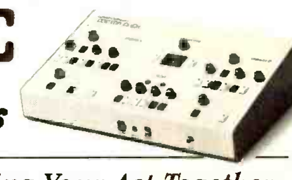
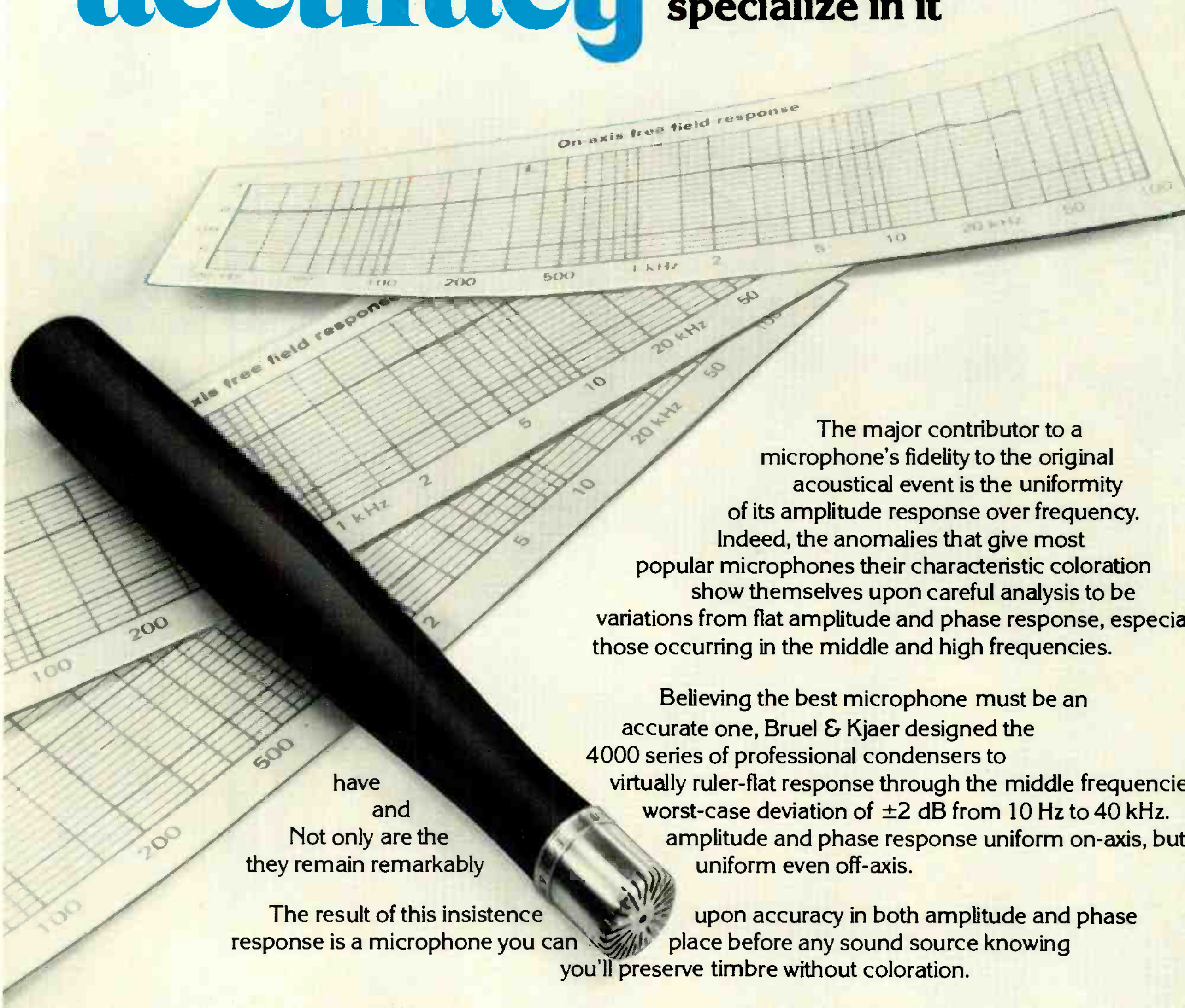


For additional information circle #68

A few words on microphone accuracy

from the people who specialize in it



The major contributor to a microphone's fidelity to the original acoustical event is the uniformity of its amplitude response over frequency. Indeed, the anomalies that give most popular microphones their characteristic coloration show themselves upon careful analysis to be variations from flat amplitude and phase response, especially those occurring in the middle and high frequencies.

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ANALYSIS OF A SESSION

Talking with Kevin Clark, Engineer for Manhattan Transfer's *Bodies and Souls* Album

"In the beginning of the *Bodies and Souls* project," Clark recalls, " the members of Manhattan Transfer [Janis Siegel, Cheryl Bentyne, Tim Hauser, and Alan Paul] wanted to do the vocals the way they were used to doing them, which was recording all the voices with U-87s. After some discussion, and AB-ing of my choices against the U-87, we decided on the following microphones, which varied according to the specific application. For lead parts on contemporary, pop pieces, I almost always used Neumann U-89s on the girls as soloists; Telefunken 251s worked occasionally. Neumann U-67s worked well on lighter pieces, interchanged a couple of times with a Neumann U-47. Backgrounds were almost always 251s, but I did use some U-67s once in a while.

"Because the vocals were recorded in various studios around Los Angeles, I carried my own vocal mikes with me to maintain a consistency of tone quality. I used the same Neumanns [47s, 89s, and 67s] in all the different rooms. The Telefunken 251s belong to United Western [Hollywood], and I used those while I was working there."

Production and Recording Techniques

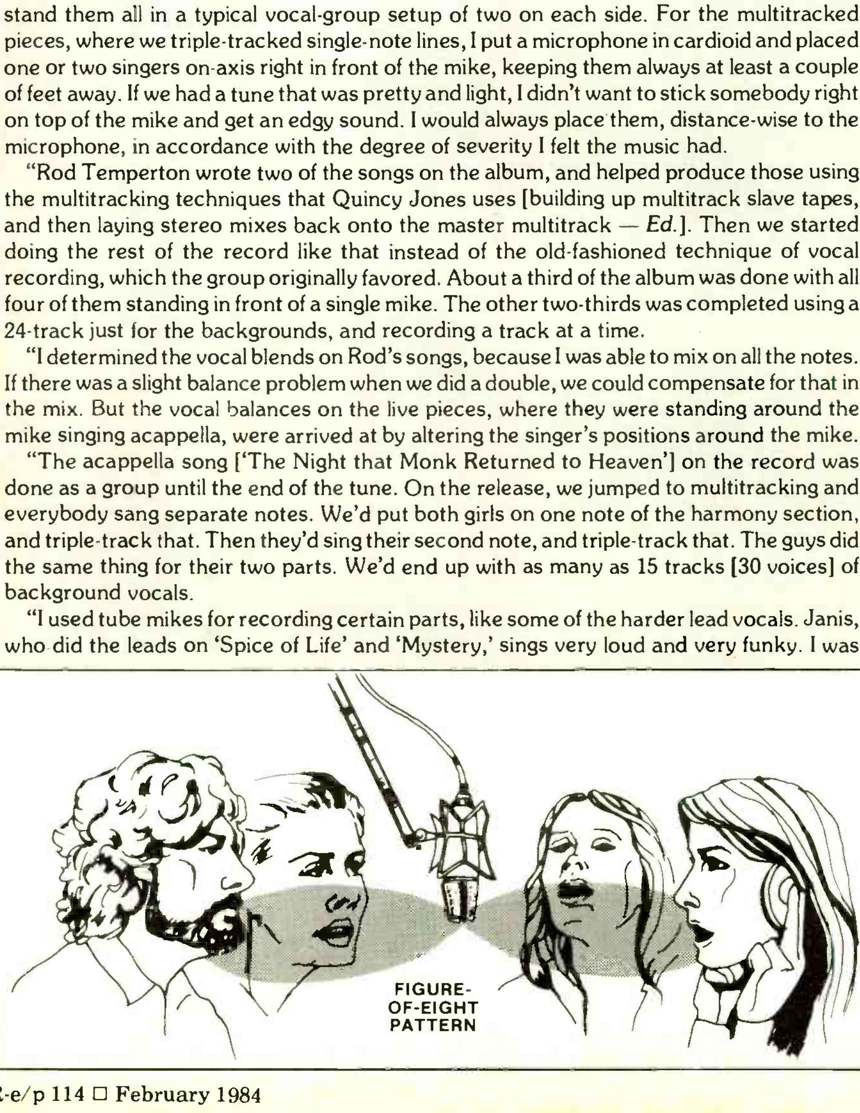
"Some tracks were very simple to record. Cheryl's vocal on 'Good-bye Love' was live with a Fender Rhodes. We kept the complete take; there wasn't one punch-in. When they sang group parts, a lot of times I would use a microphone in a figure-of-eight pattern, and stand them all in a typical vocal-group setup of two on each side. For the multitracked pieces, where we triple-tracked single-note lines, I put a microphone in cardioid and placed one or two singers on-axis right in front of the mike, keeping them always at least a couple of feet away. If we had a tune that was pretty and light, I didn't want to stick somebody right on top of the mike and get an edgy sound. I would always place them, distance-wise to the microphone, in accordance with the degree of severity I felt the music had.

"Rod Temperton wrote two of the songs on the album, and helped produce those using the multitracking techniques that Quincy Jones uses [building up multitrack slave tapes, and then laying stereo mixes back onto the master multitrack — *Ed.*]. Then we started doing the rest of the record like that instead of the old-fashioned technique of vocal recording, which the group originally favored. About a third of the album was done with all four of them standing in front of a single mike. The other two-thirds was completed using a 24-track just for the backgrounds, and recording a track at a time.

"I determined the vocal blends on Rod's songs, because I was able to mix on all the notes. If there was a slight balance problem when we did a double, we could compensate for that in the mix. But the vocal balances on the live pieces, where they were standing around the mike singing acappella, were arrived at by altering the singer's positions around the mike.

"The acappella song ['The Night that Monk Returned to Heaven'] on the record was done as a group until the end of the tune. On the release, we jumped to multitracking and everybody sang separate notes. We'd put both girls on one note of the harmony section, and triple-track that. Then they'd sing their second note, and triple-track that. The guys did the same thing for their two parts. We'd end up with as many as 15 tracks [30 voices] of background vocals.

"I used tube mikes for recording certain parts, like some of the harder lead vocals. Janis, who did the leads on 'Spice of Life' and 'Mystery,' sings very loud and very funky. I was



tertiary. You really concentrate on what makes those feelings"

The intimate working relationship that develops between the artist and an engineer during overdubs expands the traditional boundaries of the audio professional to encompass some of the tasks normally associated with the title of producer. These tasks include evaluating significant, yet intangible, aspects of a performance that are difficult to articulate. In the following section vocal teacher Warren Barigian attempts to explain the mechanics behind that elusive difference between a good performance, and a *great* one.

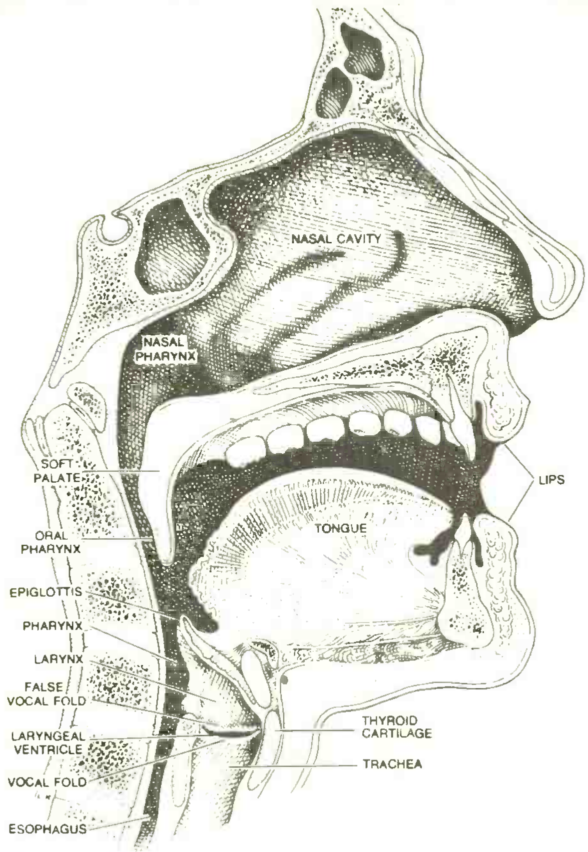
Music Carries the Singers

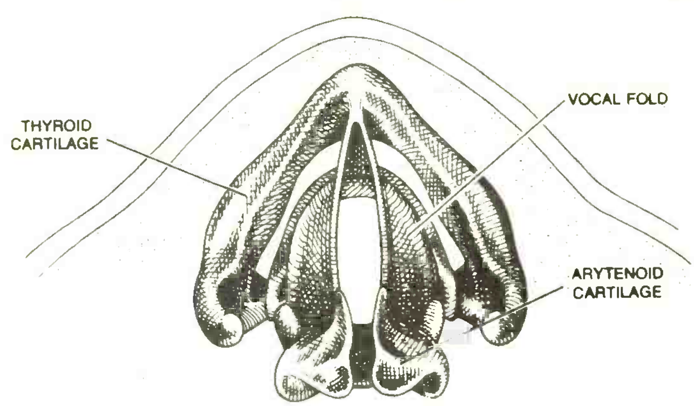
Barigian is of the belief that, generally speaking, the music portion of most contemporary records is more significant than the vocal performance. "A lot of credit for some artists' successes must be given to the engineers, the technology, the musicians and the commercial aspects of their songs," he considers. "Actually, technology can't *make* a fine artist; it can only *help* that person to sound much more significant than they are. But that still doesn't compensate for the lack of fine artistry."

Too many singers drift in and out of the song, he offers, and don't understand how commanding they could be, and how uncommanding they are. To really dramatize that point, ask yourself: Can any one of 10 singers do the song? If so, the person probably isn't a vocal artist. "A singer must be so exceptional that nobody can replace him," Barigian states. "His character and exceptionality have to come right through, which underscores the big difference between being a singer and being a vocal artist."

"Singers" are not capable of bringing alive the artistic capacity — a fact that can be demonstrated by asking a singer to deliver the lyrics without access to the music. In most cases, the rendition will be insignificant, impotent, ineffectual, boring to the point of not being able to not hold your attention. To overcome that weakness, the singer must not say the first word without knowing their "intent," which may be described as emotional direction, motivation, or purpose. "That intent must be held throughout the entire performance," says Barigian, "and includes every small word like 'a' and 'the' and even the [musical] rests, because silence doesn't have to be inert energy. Something can be happening dynamically there, such

Gene Rice decided that a music career was for him when he got his first guitar at the age of 13 in Utica, New York. He began recording local groups in 1967, and by 1972 felt he was ready to make the move to Nashville, where he landed his first job at Fred Carter Jr.'s Nugget Studios. Since then he has worked with such country legends a Eddie Arnold, Willie Nelson, Tom T. Hall, Larry Gatlin and, of course, Grammy-award winning Alabama.





The Voice Organ is composed of the lungs, and the larynx, pharynx, mouth and nose, shown in section in the left-hand diagram. The larynx is a short tube at the base of which are twin infoldings of mucous membrane, the vocal folds. The larynx opens into the pharynx; the opening is protected during swallowing by the epiglottis. The larynx, pharynx and mouth (and in nasal sounds also the nose) constitute the vocal tract. It is a resonator whose shape — which determines vowel sounds — is modified by changes in the position of the articulators: the lips, the jaw, the tip and body of the tongue and the larynx. The vocal folds, seen from above in a transverse section in the top diagram, are opened for breathing and are closed for phonation by the pivoting arytenoid cartilages.

LINDA RONSTADT'S *WHAT'S NEW* — *continued . . .*

partitions. On the bottom were gobos [4-foot high, by 4-foot wide, by 9-inches thick, with Sonex on one side and wood on the other], and then about three feet of plexiglass running above that on each panel. We positioned the booth right between the guitarist and the piano so she could see Nelson [Riddle, the session conductor], and Don Grolnik [pianist] could see her. She was pretty well isolated even though the booth was constructed of just gobos.

"The room mike, which was positioned quite high, didn't pick up much leakage from her vocals. Only on one or two tunes when the orchestra wasn't playing at all, and she was singing very loud, could you hear the leakage. Overall, it really wasn't a problem.

"Linda generally stayed right up on the mike for a big, warm vocal sound. Rather than use the standard Neumann pop filter, we cut off the top of that filter and fabricated a custom filter of our own. She was able to put her nose into the hole of the Neumann filter and sing as close to the mike as she could get; at the most, she was two inches from the capsule.

"I don't really remember the exact EQ settings for each performance, but it was relatively flat. Maybe just a little bit of high-end. Certainly, there was nothing major. Of course, there was no doubling or technical gimmicks in the entire record. I guess the only special processing was the stimulating effect of chocolate-covered doughnuts on Linda's vocal performance."

On the Process of "Comping" Vocals

"Some vocal tracks can be recorded basically in one pass, and then touched up by dropping phrases, words, syllables, or whatever. But comping vocals is becoming fairly common with a lot of singers, because it actually can improve a good vocal, and because it's convenient. You might have an extraordinary performance that is colored by one or two phrases having mysteriously bad intonation. Replace those phrases and the performance blossoms. Phrasing is to a singer what interpretation is to Freudian psychologists. Comping can lend refinements to the phrasing, in a comparative and thereby fertile environment.

"The primary concern about combining vocal parts though, is not to lose the continuity —the feel — of the song. In fact, if one chooses the right takes to comp, the vocal-over can be made to sound *more* consistent. One way to improve the chances of getting consistent takes is to allow the artist to do as many complete vocal passes as possible at first; rather than singing one pass, listening to it, singing another, listening to it, etc. If the artist can stay in the studio and keep the same, or similar, attitude through several performances — say three or four tracks — then you can work on all of them at once, and find the best sections to make one great performance.

"At that point, if it's still not happening, an artist can go back into the studio and record three or four more passes. In a final phase the vocalist sings just a verse, or half a verse, or a chorus. The focus is to record a song that sounds like one complete thought." □□□

attention to make it come out the best it can. Recording vocals and trying to make sure that the drum parts or the synth sounds are correct is just *too* much to handle at one time. There are occasions where I'll do a live scratch vocal, while the instrumental tracks are being recorded, and that may turn out to be a great take. But I'd rather focus my *whole* attention on just the vocal track, and do that right."

When an engineer doesn't have to deal with a studio full of people, the recording environment becomes very intimate — just the vocalist in the studio, and the engineer in the control room listening through headphones to the same cue mix. In that type of environment, critical listening ability becomes an essential tool.

"To me, the recording process is becoming more listening, and less changing settings and moving faders," says engineer **George Massenburg,** who worked with Linda Ronstadt on the *What's New* sessions. "You not only look for mistakes; you must concentrate on the performance on a higher level. One stands away from the performance and listens objectively; now one listens closely for pitch; now one listens for the effect of the performance, the way it works with the music; now one listens for the little turns of phrasing that make you smile. You jot those down so that when you do a comp, you present a complete idea to the artist.

"It takes *incredible* concentration and mental hygiene. Listening to music triggers memories of feelings, because music is patterns, and therefore rather

During his 15 years as a professional singer, **Paulette McWilliams** has recorded over 200 solo commercials both in Chicago and Los Angeles for clients such as Coors, McDonalds, and the Milk Advisory Board. Her career has included a stint as one of Bette Midler's "Harlets"; lead singer with Rufus (before Chaka Khan); a duet accompaniest on Marvin Gay's recent tour; vocalist with Johnny Mathis on his last album; and vocal soloist for saxophonist John Klemmer.

Dr. Lloyd Stenbeck's credentials span a wide range of disciplines, from nutrition to chemistry to chiropractic. He is currently a fellow at the Royal Society of Medicine (London), fellow at the American Institute of Chemists, member of the International Academy of Preventative Medicine, associate member of the International College of Applied Nutrition, and certified by the American College of Chiropractic Orthopedists. He is most noted for his work with Olympic athletes, and professional vocalists.

As a vocal coach in Los Angeles, **Warren Barigian** began to notice the subtle but direct relationships between the voice, and various conditions of the body. During the course of almost two decades, he has collated his dramatic breakthrough into a system of study that is said to be able to increase the dynamic and frequency range of the human voice, thus providing the singer with a greater potential for artistic expression. In fact, with the help of psychiatrist, psychologists, nutritionists, and various other professionals, Barigian says that he can give a "tone deaf" person a professional quality signing voice in three months. His clients include Jackson Browne, various members of the Grateful Dead, Cher, Bonnie Raitt, Kenny Loggins, and many more.

These facts are not presented to lure an audio professional to the field of medicine or therapy. Their inclusion emphasizes the opportunity/responsibility that the engineer possesses to direct the outcome of a vocal performance. The environment and attitude that an engineer creates in the studio become more obvious when working with a vocalist, as opposed to recording a band of instrumentalists. Where musicians' feelings and general level of health are filtered through their instruments, and obscured by mechanical or electronic processes, the influences on a vocalist go directly to the throat, through the mike, and onto the tape. In addition, the physical strain of pushing beyond common-sense limits can actually cause irreparable damage to the vocal instrument.

Of course, an engineer's primary objective is to record the artist in the best way he or she can. But these peripheral issues are inescapable — they are part of the turf of which experienced engineers inevitably become aware, whether consciously or unconsciously. To ensure that a session runs as smoothly and efficiently as possible, and allow the artist to unleash his or her full potential, the engineer/producer may wish to explore several options.

Paulette McWilliams, an LA session vocalist, mentions a couple of areas where the environment affects her performance: "Most singers don't like the temperature in the room to be too cold or too warm. When it's too cold, the throat closes up, and strains. Excessive heat doesn't bother the voice as much; it just makes the studio uncomfortable, which also affects the way you do a song."

Studio lighting, too, is important, she considers. "The lighting helps create the right mood. Sometimes it's not always what you'd like, and you just learn to be adaptable to the available conditions. But if the studio can give you the right atmosphere — and in most studios they can — then it helps a lot for inspiration on the solo performances. Again, there has to be a balance; if it's too dark, you get sleepy and the performance suffers, too."

Vocalist's Responsibilities

Engineer **Gary Skardina**, who also sings, writes hit songs, and owns Music Grinder, a 24-track studio in Los Angeles, points out that a vocal performance can be very demanding for a soloist. "Even though there is only one fader up on the console, there's a lot going on to make that track happen," he offers. "Every second of the performance, the artist is concerned with remembering the lyrics; the delivery; the notes; the pitch; whether they should keep the vibrato down or not; how they should close certain syllables; how to avoid a problem singing any type of consonance . . . the list goes on.

"And, on top of that, they have to deal with the recording aspect — how close they are to the mike at any one time. If they sing something very loud, such as a scat, they may have to turn their head off the mike. Then, if the engineer or producer changes something, like a word or a new phrasing suggestion, the singing can get pretty confusing."

To complement this high level of sophistication demanded in today's music, the recording industry has evolved the concept of the vocal specialist — an engineer who primarily records vocal tracks in an overdub situation. "The lead vocal track is extremely important to a record," notes Skardina, "and deserves a great deal of time and

ANALYSIS OF A SESSION

Talking with George Massenburg, Engineer for Linda Ronstadt's *What's New* Album

"Very often engineers approach a project such as this," Massenburg considers, "with the idea of capturing 'a sound' that may be reminiscent of a particular era, and choose microphones accordingly. On the *What's New* project we found that the most important considerations for reproducing that genre of music were the arangement and the vocal phrasing, as opposed to the engineering. We first referred to recordings from the Forties and Fifties; as an example, listen to Frank Sinatra's "Only the Lonely" record. After listening, you're struck by the fact that you haven't really *heard* individual horn or string parts, because they blend very well. That's certainly a function of arrangements and vocal performances, rather than the recording.

"On the back of Linda's album is a credit to [engineer] John Neal, who was really a great help with the project. He's done hundreds of setups like this for television and films [at Hollywood Palace, 20th Century Fox, and Warner Brothers]. We used a basic orchestral miking approach [at the Complex Studios, West LA] to capitalize on the leakage from section to section. The overall pickup came from one good M-S mike [a modified AKG C24] that was very carefully placed, and whose dimension was very carefully adjusted. Then we filled in with tight mikes — including three on drums [one on the snare that wasn't used, and two overheads].

"If you do this type of setup correctly, the actual balances are usually static. Like the Sinatra recording, all the dynamics and movement take place via the arrangements (which were done so well by Nelson Riddle) and vocal dynamics. A lot of what happened in the sessions was really due to the quality of the arrangements and conducting."

Vocal Microphone Selection

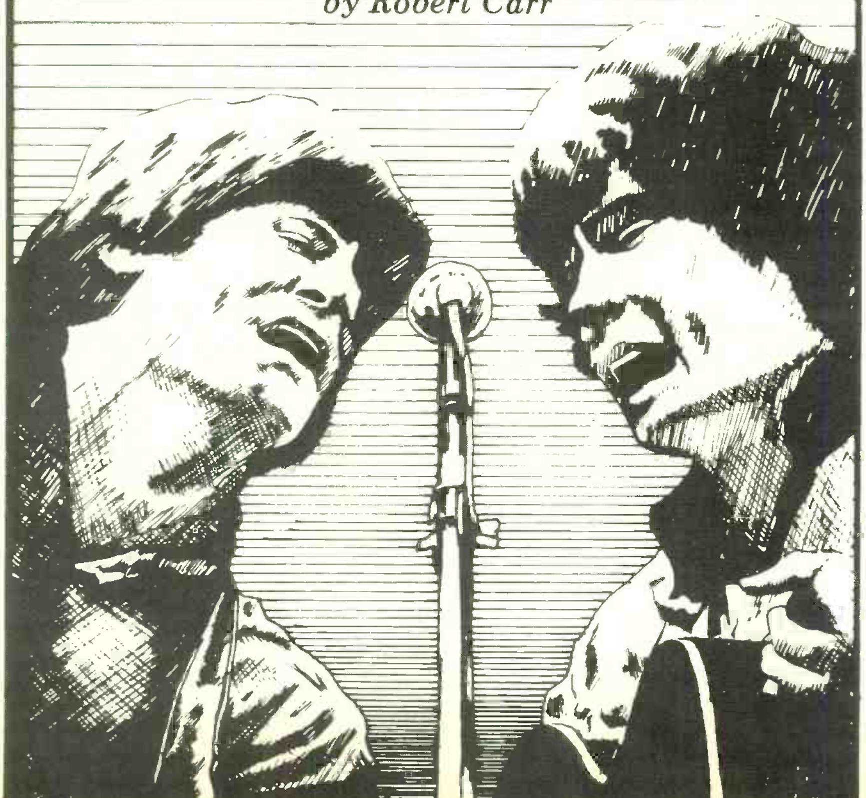
"Linda's vocal microphone was a Neumann U-67, which I've had for about 19 years; it's the same type of mike that Val Garay used for all those great rock-and-roll records. She's used to working a mike in a particular way [with her nose in the Neumann pop filter], and there was no need to change. The vocal tracks were cut live with the orchestra, with a few overdubs and some punch-ins to patch up the phrasing.

To get some isolation on Linda's vocal track, we built a little booth out of seven-foot-high

Production Techniques

RECORDING VOCAL AND HARMONY SECTIONS

by Robert Carr



Although many musicians look as though they were born attached to their axe, the voice is still the only musical instrument that is *truly* a part of the human body. This whimsical, and possibly mundane, point underscores a great deal of truth. The human body is a living organism and, for that reason, is in a constant state of flux from all sorts of environmental, emotional, mental, and even spiritual influences. The only constant upon which an engineer can depend is that from one session to the next, regardless of the safeguards one tries to implement as a deterent or stabilizing force, there will always be fluctuations in frequency range, timbre, power, emotional content, and a host of other vocal parameters.

As a cursory introduction to how the vocal mechanism operates, the reader should refer to the accompanying figure and brief explanation reprinted with permission from Scientific American's *The Physics of Music*. For a more in-depth study on this aspect of the voice, the research paper in which the drawing first appeared (entitled: "The Acoustics of the Singing Voice," by Johan Sundberg) contains a wealth of scientifically derived information.

This article, as the others in *R-e/p's* continuing series on recording various instruments have attempted, will explore several peripheral areas that should be of interest to the engineer faced with the dilemma of achieving a premium vocal recording. A variety of miking approaches have been abstracted from the main text, and presented as sidebars for easier access to specific topics.

Factors Influencing Vocal Parameters

As mentioned earlier, the vocal organ forms an integral part of the body. Consequently, any seemingly unrelated disorder in the human system manifests itself through the voice. Anger or fear, for example, can impair vocal sounds to the point of nullifying speech. [As explained in greater detail in the article "dBs Can be Hazardous to Your Health," by Martin Polon; *R-e/p*, October 1979 issue — *Ed.*] Joy and inspiration, on the other hand, strengthen spoken or sung words. Vocal stress tests are becoming more and more popular with law enforcement as a means of reading tension and separating truth from lies.

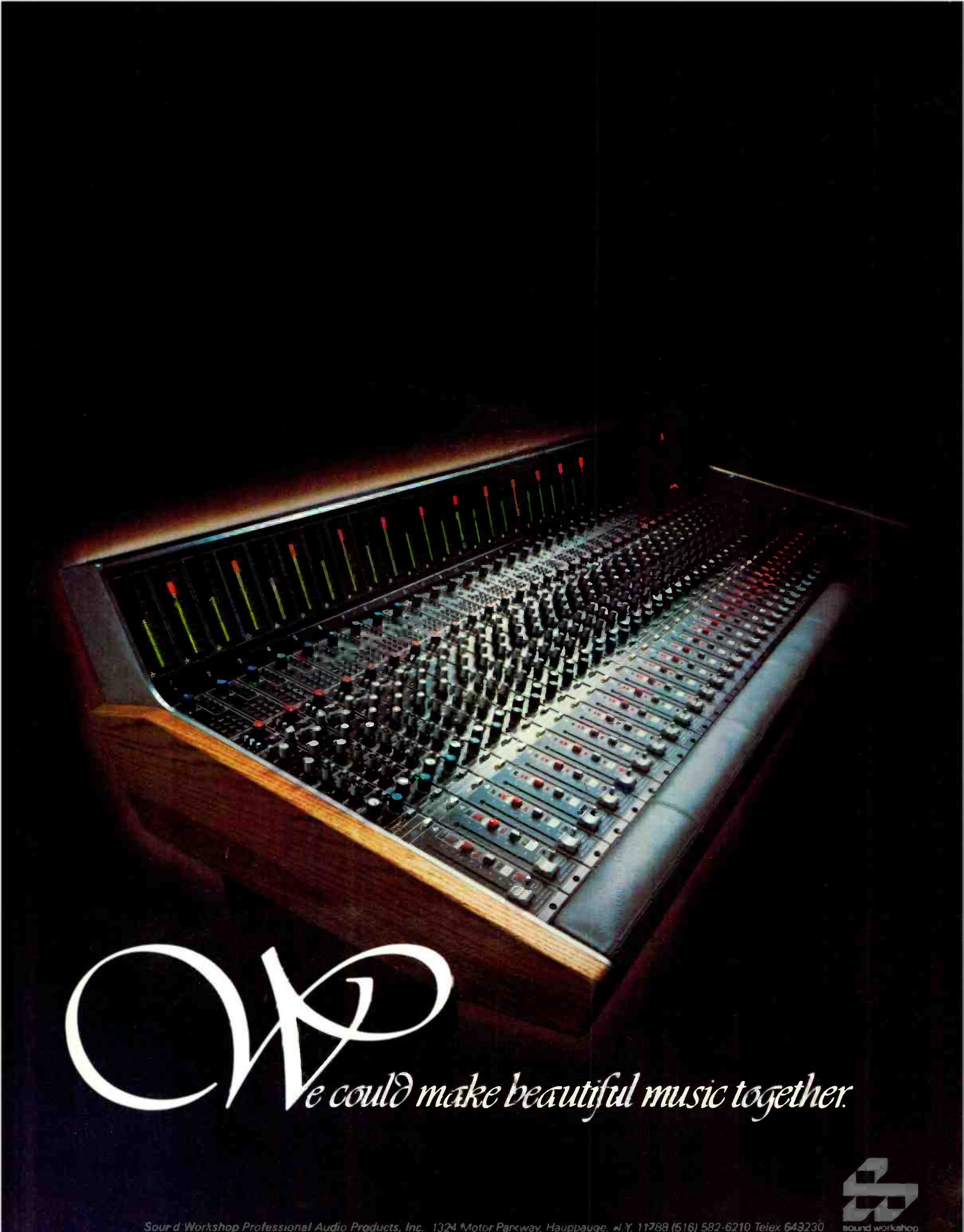
In addition, the voice is extremely sensitive to imbalances in the normal bodily functions. Los Angeles-based vocal teacher **Warren Barigian** has attributed vocal fluctuations and soreness to such diverse factors as too-high or too-low a level of hydrochloric acid in the stomach (hydrochloric acid aids in digestion), food allergies, weak adrenal glands, jet lag, elevation changes, seasonal changes, recreational drugs and alcohol, and any combination of the above.

"By far the biggest problem that singers have to deal with is stress," points out **Dr. Lloyd Stenbeck,** an orthopedic chiropractor who specializes in nutrition, and who has helped numerous vocalists in developing their voice. "Not only as a result of worrying about doing a good performance, but also the pressures associated with a poor lifestyle of erratic hours on the road, exposure to excessive amounts of cigarette smoke, alcohol and drugs, and subconscious-stress emotions that people hold in their throats."

Most engineers also are exposed to these same pressures. As a point of reference for use by vocalists and technicians alike in those desperate studio situations, we offer here a couple of Dr. Stenbeck's simple suggestions for dealing with the above mentioned causes of stress. Two tablets once a day of a nutritional supplement called Pineal Protomorphogen, manufactured by Nutridyne, and which is commonly used to strengthen the adrenal glands of people that have been under a great deal of pressure, seems to have a hormonal affect on the vocal cords as well, and may enhance someone's singing potential.

A quick, effective concoction that detoxifies the bloodstream, the brain, and the central nervous system after too much smoke, drugs or alcohol, Stenbeck offers, is a tea made from any three of the following herbs combined in a 1:1:1 ratio: Safflower, Damiana, Elder Blossoms, Foti, Skullcap, and Black Cohash. A fourth herb like Licorice, Spearmint, or Peppermint may be added for flavoring. The addition of Valarian root is also helpful in counteracting stress, and may be included on a 1:1 basis. Beside cleaning and soothing the central nervous system, a few cups of this strong tea should show marked results in vocal quality.

. . . continued overleaf —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tering deck (scheduled for possible replacement earlier this year with a new Soundcraft Series 20 ¼-inch stereo machine), Lexicon 224 digital reverb and effects processor, Eventide H910 Harmonizer, UREI LN1176 compressor-limiter, Rebis noise gates and analog delay modules, plus Drawmer DS20 dual gates and DL221 stereo compressor-limiter. A spacious recording area, complete with a home-brewed acoustics treatment, can easily accommodate up to 24 musicians — in the past the studio has been used to track large brass bands — or 40-plus choirs and vocal ensembles.

In another room in the studio can be found a real-time cassette duplication area, comprising a bank of 27 Nakamichi 581 and 582 decks for recording, and a ReVox B77 two-track reel-to-reel and Nakamichi 480Z for replay.

Gateway's session business is split between high-quality demos and album dates. With a tightening of recording budgets — not to mention self-financed or spec projects — Ward reports that it is not uncommon for bands to be looking to spend between £3,000 and £4,000 (around $6,500) on an album, including musicians' fees. "There has been a trend," he observes, "towards cutting a complete album in just a week — including a day to remix it! We've even had albums that had been done in nine hours: two, three-hour sessions for tracking, and a three-hour session for mixing!

"Regarding the fate of tapes that leave the studio, we handle a lot of specialist releases — including ethnic music recording — plus albums or EPs that will be of local interest, or can be sold by bands at gigs around the country, particularly on the folk-music circuit."

But rather than simply take any session that comes its way, Gateway looks for a positive involvement with the music being produced, Ward says. " We've made a conscious move towards quality work that is interesting to record, and with which we can become involved on a creative level." And Ward, speaking from bitter experience, in particular having to rush through sessions and then discover that the band doesn't like the sound, added that Gateway's crew "would rather have the studio empty, than have a bunch of idiots in, and have to worry about the mikes being lost!"

If the key to success in today's changing recording industry is towards diversification — audio-for-video, scoring, commercials production, or film post-production being some of the more familiar areas for lateral development — the studio has come up with a rather interesting concept. Recognizing that there were no official educational courses being offered in Britain for low-level users of recording hardware, last September saw the setting up of Gateway's own in-house training school. At the time of my visit the courses were being held every two weeks in a couple of upstairs rooms, with a maximum of a dozen people at each session.

"So far the response has been excellent," Ward enthuses. "The three-day course is designed for the person looking to invest around £2,000 ($3,000) in a home studio based on a TEAC Portastudio, Fostex Multitracker, or Fostex/Otari/Tascam four- or eight-tracks. But we don't offer the course for people that want to make a career in the recording industry; instead it has been configured more towards letting people become familiar with the equipment — how to hook it all together, and get the best results from it all.

"Currently the school is equipped with a Fostex A-8 [eight-track on quarter-inch], Fostex mixer, and various peripherals. We plan to extend the courses to include computer mixing, and synthesizer patches and set ups. There is even the possibility of it developing into a road show that we could tour around the country, possibly in cooperation with various pro-audio dealers." □□□

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TOWNHOUSE STUDIOS, 1 Godolphin Road, London W12. 01-743 9313.
LANSDOWNE STUDIOS, 2 Lansdowne House, Lansdowne Road, London W11. 01-727 0041.
GATEWAY STUDIOS, 1A Salcott Road, London W11. 01-223 8901.

For additional information circle #63

STUDIO UPDATE

FACILITIES
EQUIPMENT
PEOPLE

Gateway Studio owner Dave Ward.

Gateway's real-time cassette duplication room.

equipped with a non-SSL console, simply because we should be able to provide a potential client with a choice of recording hardware. But at present nothing else comes close to the quality and facilities offered by an SSL desk."

Lansdowne Studios, located in Holland Park, West London, and this year celebrating its 25th anniversary, is co-owned by producer/-engineer Adrian Kerridge, and composer Johnny Pearson. Having established an enviable reputation for audio recording and production, a couple of years ago Kerridge and studio manager Chris Dibble made the conscious decision to move into the expanding field of high-quality audio-for-video, scoring, and film sound post-production. And the timing of that decision couldn't have been more auspicious, Kerridge conceded, because he was able to put together a cost-effective facility based on ¾-inch U-Matic videocassette, rather than 35mm mag and film transports. "Video," he says, "allows us to get 20% more music in the can during the same time period than when working with film."

For scoring or sweetening to picture, Lansdowne's main control room is fitted with a 7- by 5-foot video screen and Barco projection system. To ensure that there are no speed anomolies when working with videocassette copies of visuals shot originally on film, film-to-tape transfers are made at an outside facility equipped with a rotating-prism telecine machine capable of running at 25 frames-per-second, rather than the normal 24 FPS film projection rate. (It should be remembered that England, like the rest of Europe, is referenced to an EBU 25 FPS video frame rate and 50 Hz power frequency; and, unlike the USA's 30/29.97 FPS non-drop and drop-frame rates, also utilizes a common frame rate for monochrome and color broadcast and video recording.) In this way, Kerridge points out, each video frame is uniquely and correctly indexed against the corresponding SMPTE vertical interval timecode location laid down on the U-Matic slave tape.

Kerridge is also extremely excited about a new computer-controlled system developed by The Music Design Group, Hollywood, that provides visual cues for the conductor and producer during scoring dates. The VideoScore system, scheduled for delivery at Lansdowne in May, enables video-generated equivalents of the film-style "streamers" and "punches" to be superimposed on video monitors located in the studio, and on the control-room projection screen. (In conventional film scoring, individual frames of a work print would be treated with punched holes to designate tempos and beat counts, while diagonal streamers marked down the length of film would show during projection as a vertical line that moves from left to right on the screen, and might be used to signify the imminent arrival of a scene change, or a dramatic punctuation in the film's action.)

The location and spacing of these "electronic" streams and punches can be marked on the fly from a companion typewriter-style keyboard, tallied from SMPTE timecode locations, or entered in feet and frames; internal conversion from hours/minutes/seconds/frames to feet/frames, or vice versa, is carried out automatically. Printouts can even be annotated for the composer, musicians, and engineer. In addition, complex changes can be made to the timings and cue locations without the need to reconfigure the videotape; the unit simply recalculates the revelant data and intervals. Planned enhancements for the system include an electonically generated clock that can be superimposed on the video picture for accurate countdowns to musical cues.

After scoring or sweetening to picture on a SMPTE interlocked multitrack, the final master format that leaves Lansdowne depends on the nature of the project. For conventional film-style dubs, the studio uses an Albrecht 35mm three-stripe dubber, while for audio material destined for layback to videotape Lansdowne — in keeping with a growing number of UK post-production rooms and broadcasters — dubs and the mixes to eight-track on one-inch. Having been supplied with an eight-track tape that already contains, for example, effects, dialog and a rough trial mix (plus SMPTE timecode and video sync), additional stereo music tracks can then be added, and this first-generation tape used during the final mono or stereo mix and layback to videotape.

Control room equipment centers around a 36-input Cadac console with transformerless mike pre-amps, a Melkuist GT800 disk-based automation package, Studer A80 24-, eight, and a pair of two-track decks, an Audio Kinetics Q.Lock 3.10 SMPTE synchronizer, and a custom Cadac monitoring system. (Incidentally, Cadac was part-owned by Adrian Kerridge and console designer Clive Green until its unfortunate demise several years ago; Green is still active with custom console design and manufacture.) A second, smaller room is available at Lansdowne for post-production, and 35mm mag and ¼-inch transfers.

Currently, Kerridge reports, Lansdowne spends 50% of session time working on audio-for-video, of which the majority (some 70%) is for television projects, and the remainder film scoring and TV jingles. Conventional music sessions range from orchestral dates, jazz, and radio jingles, to string and horn overdubs.

"The key to a successful future in the recording industry," Kerridge considers, "is diversification, and the ability to produce high-quality audio for *any* medium. To a certain extent, I think it's possible to compare the problems facing certain [U.K.] studios, many of whom are finding it hard to adapt to changes affecting the business, with the medical profession. Consider the analogy of the doctor that graduated 25 years ago, but who hasn't been following the medical literature over the intervening years, with one who has kept himself abreast of the changes in modern drugs and treatments. Obviously the latter has the same information at his disposal on current techniques as a recently qualified doctor but, in addition, he has a wealth of day-to-day experience and maturity. Lansdowne prides itself on not only an excellent track record, but as a studio that has kept up with the changing requirements of our industry."

Moving onto the smaller-studio market, Gateway Studios, situated in the Clapham district of South London, and co-owned by musician/engineer Dave Ward, and Lisa and Rory Monck, over the last six years has grown from an eight-track on half-inch, Tascam-equipped room to its present 16-track configuration. Present recording hardware includes a Soundcraft 1624 board and SCM-381 two-inch multitrack equipped with BEL noise reduction (which, being based on a 2:1 compansion ratio, is said to be compatible with dbx NR units), Tannoy Super Red monitors, Ampex ATR-700 mas-

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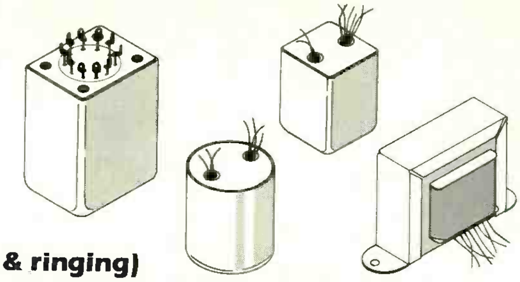

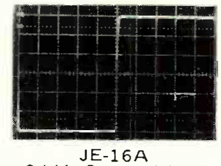

JE-16A
2 kHz Square Wave

INPUT TRANSFORMERS AND SPECIAL TYPES

Model	Application	Impedance Ratio Pri-Sec	Turns Ratio Pri:Sec	20 Hz Max Input Level[1] (dBu)	Typical THD Below Saturation (%) 20Hz/1kHz	Frequency Response (dB ref. 1 kHz) 20Hz/20kHz	Band-Width[2] –3dB @(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	170	–10	<1.75	1.7	–30	1	A=1 B=2	63.61 68.25	42.49 45.60	29.32 31.46
JE-13K7-A JE-13K7-B	Mic in for 990 or I.C.	150-3750	1:5	+8	0.036/0.003	–0.10/–0.22	85	–20	<3	2.3	–30	1	A=1 B=2	63.61 68.25	42.49 45.60	29.32 31.46
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	41.48	27.72	21.65
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	102.86	68.72	47.42
JE-11P-1	Line in	15K-15K	1:1	+17	0.045/0.003	–0.03/–0.25	85	–23	<1		–30	1	3	39.53	26.41	20.62
JE-6110K-B JE-6110K-BB	Line in bridging	30K-1800 (10K-600)	4:1	+24	0.005/0.002	–0.10/–0.30	75	–15	<1		–30	1	B=1 BB=2	62.31 70.95	41.63 47.38	30.56 32.70
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	40.98	27.37	18.89
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	151.90	101.47	70.01
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	78.62	52.52	36.24
SPECIAL TYPES																
JE-MB-C	2-way[3] mic split	150-150	1:1	–2	0.180/0.005	–0.25/–0.20	88	–15	<1		–30	2	3	34.08	22.78	17.78
JE-MB-D	3-way[3] mic split	150-150-150	1:1:1	–2	0.180/0.005	–0.25/–0.16	100	–12	<1		–30	3	3	59.57	39.80	31.08
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	96.29	64.32	44.38
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	43.04	28.76	22.46

1. 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 =	1 5/16" Diam.		× 1 9/16"
2 =	1 3/16"	× 1 3/16"	× 1 5/8"
3 =	1 1/8" Diam.		× 1 1/16"
4 =	1 1/2"	× 1 3/4"	2 1/2" w/solder terminals
5 =	1 5/8" Diam.		× 1 3/4"
6 =	1 1/8" Diam.		× 15/16"

NICKEL CORE OUTPUT TRANSFORMERS[6]

Model	Construction	Nominal Impedance Ratio Pri-Sec	Turns Ratio Pri:Sec	20Hz Max Output Level[7] (dBu)	across (n) windings	600Ω Termi-nation Loss (dB)	DC Resistance per Winding (Ohm)	Typical THD Below Saturation (%) 20Hz/1kHz	Frequency Response (dB ref. 1kHz) 20Hz/20kHz	Band-Width –3dB @ (kHz)	20 kHz Phase Response (degrees)	Over-Shoot[8] (%)	Package[9]	PRICES 1-19	100-249	1000
JE-123-BMCF	Quadfilar 80% nickel	600-600 150-600	1:1 1:2	+28	2	–1.1	20	0.002/0.002	–0.02/–0.02	>450 158	–2.1 –4.1	<1	7	87.41	44.17	30.47
JE-123-DMCF	Quadfilar 80% nickel	600-600 150-600	1:1 1:2	+21	2	–1.0	19	0.004/0.002	–0.02/–0.00	>450 230	–1.2 –2.5	<1	8	50.71	33.88	23.38
JE-123-BLCF	Quadfilar	600-600 150-600	1:1 1:2	+32	2	–1.1	20	0.041/0.003	–0.02/–0.01	>450 168	–1.9 –4.0	<1	7	61.30	35.79	24.70
JE-123-DLCF	Quadfilar	600-600 150-600	1:1 1:2	+27	2	–1.0	19	0.065/0.003	–0.02/–0.01	>450 245	–1.2 –2.5	<1	8	39.61	26.45	19.42
JE-123-SLCF	Quadfilar	600-600 150-600	1:1 1:2	+23.5	2	–1.1	20	0.088/0.003	–0.03/–0.01	>450 245	–1.2 –2.8	<1	9	33.48	22.35	15.43
JE-112-LCF	Quadfilar	600-600 150-600	1:1 1:2	+20.4	2	–1.6	29	0.114/0.003	–0.03/–0.01	>450 205	–1.2 –3.2	<1	10	25.48	17.01	12.49
JE-123-ALCF	Quadfilar	66.7-600	1:3	+26.5	3	–1.3	8	0.125/0.003	–0.04/+0.06	190	–4.6	<6	8	42.14	28.15	19.42
JE-11S-LCF	Bifilar w/ split pri.	600-600 150-600	1:1 1:2	+30	1 (sec)	–1.7	63	0.058/0.002	–0.02/+0.01 –0.02/–0.05	>10MHz 155	+1.1 –4.1	<1	8	42.14	28.15	19.42

6. Multifilar construction has no faraday shield. All specifications are for 0Ω source, 600Ω load.
7. Max output level = 1% THD; dBu = dBv ref. 0.775 V
8. Source amplifier –3dB @ 100kHz
9. Output transformers are horizontal channel frame type with wire leads, vertical channel frames available.

PACKAGE DIMENSIONS:

	W	L	H	Mounting Centers
7 =	1 1/2"	× 2 5/16"	× 1 15/16"	2 13/16"
8 =	1 5/16"	× 1 15/16"	× 1 5/8"	2 3/8"
9 =	1 1/8"	× 1 11/16"	× 1 3/8"	2"
10 =	1 1/16"	× 1 7/16"	× 1 3/16"	1 3/4"

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These charts include the most popular types which are usually available from stock. Many other types are available from stock or custom designs for OEM orders of 100 pieces or more can be made to order. Certified computer testing is available for OEM orders. Call or write for applications assistance and/or detailed data sheets on individual models.

Prices shown are effective 2/1/84 and are subject to change without notice.

For additional information circle #62

Helios board will be removed from Studio One during March or April of this year, and is destined to replace the vintage Neve console currently gracing Ramport's control room; a new 48-input Solid State Logic 6000E audio/video production and recording console is planned for installation in One. Townhouse Three will also undergo a certain amount of refurbishment, including the sorting out of one or two acoustic problems, and then will be hired out as a mid-priced facility. (Studio rates are expected to be in the region of £50 per hour; around $70 to $75.)

Meanwhile, earlier this year the original SSL 4000B desk in Studio Two (and one of the first Solid State consoles sold to a U.K. studio) was destined to be replaced by a brand-new 4000E equipped with a Total Recall automation system. (Incidentally, The Manor was re-equipped some 18 months ago with an SSL 4000E console, with plans to add Total Recall capability in the near future.)

Douglas also reports that the studio has been increasing its complement of digital hardware. In addition to a 3M digital four-track machine for disk-mastering duties (and shared on a cooperative basis with Bronze Records' Roundhouse Studios, which facility also owns a 32-track machine for tracking dates), the complex has purchased Sony PCM-1610 and PCM-F1 audio processors — the latter being referred to by Douglas as a "rich producer's ReVox." But high-end appeal notwithstanding, he adds, the F1 enables studio staff to keep itself abreast of current digital technology, and provides a cheaper and easier format for making backup tapes, and copies for the band. While the studio is currently considering the possible purchase of a Sony PCM-3324 digital multitrack, Douglas reports that because of compatibility considerations — and the small number of DASH-format machines in the UK — sessions would be run in parallel to a Studer A800 analog machine.

And talking of exotic recording hardware, until about a year ago Townhouse was the owner of one of the very few Telefunken M15 32-track on two-inch machines that found their way into an English studio. Problems of format compatibility meant that few producers elected to track on the M15 — Douglas can recall it being used on no more than 10 projects during the four-year period that the studio owned the deck — although he remembers that the machine's electronic performance was very good indeed. Because of the M15's interleaving track layout, it was possible, he says, to add another five tracks to a full 24-track tape; indeed, on several occasions overdubs were added in this way to save the hassle of locking up an additional 16- or 24-track.


Lansdowne's Cadac-equipped audio-video room: Adrian Kerridge (left), and studio manager Chris Dibble.

Regarding Townhouse's decision to equip all three studios with SSL consoles, Douglas offers that "with the possible exception of the new Neve DSP [digital] console, nobody else is making consoles worth having. However, I would like to have one of the rooms

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STUDIO UPDATE

FACILITIES
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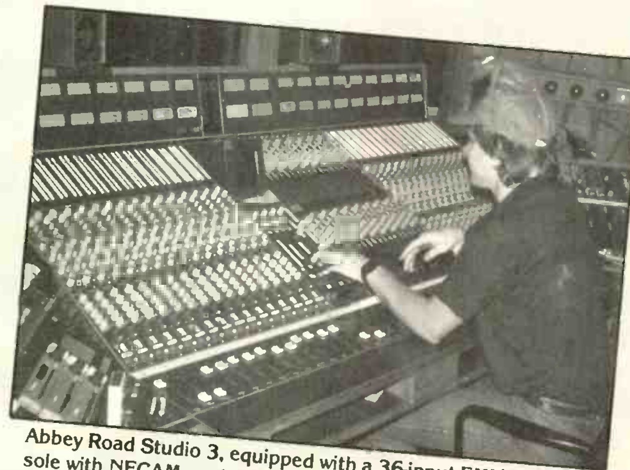
EMI-Abbey Road Studio 2: operations manager Peter Vince (left) and Eric Wolfson, co-member of the Alan Parsons Project.



Abbey Road Studio 3, equipped with a 36-input EMI/Neve console with NECAM, and Studer A800 24-track.

multitracks and mastering machines), and a modified JBL monitoring system with EMI-designed crossovers and Yamaha P2201 power amps.

Control-room acoustics were designed by Peter Dix of EMI's Central Laboratories, the intention being, Vince says, to produce "a plain, functional look to the room. We were not after a 'flat sound' in the room, but rather an environment where you could listen under similar conditions to those found in the 'average' living room. For example, in one of our other studios [The Penthouse] we first flush-mounted the monitors in soffits in the front wall, and then used third-octave EQ to tune the room. The result, however, was a *lousy* sound. Instead, we opted to mount the speakers in the room [on tall stands, or suspended from the ceiling by means of cradles and chains]. The result was far better than we had before. By mounting the monitors in the room itself, we can more closely duplicate the type of monitoring layout people have in their homes. We have found that the sound [of tapes mixed on such monitoring systems] will carry from space to space, rather than only sound good in that particular environment."

One unusual feature that caught my attention was a system of metal shutters set in the wall beside the observation window that looks down into Studio 2's recording area. It turns out that London has particularly stringent fire-prevention regulations, and that such shutters had to be fitted (at not inconsiderable expense, I discovered) to meet the relevant building safety codes.

Vince says that with the increasing popularity of Compact Disc players in the U.K., the studio is now handling more digital mastering, and hires the equipment on an "as-needed" basis: Sony PCM-1610 audio processors and companion U-Matic videocassette decks, 3M 32- and four-track Digital Mastering Systems, and Soundstream eight-track machines (the latter mainly for classical sessions).

At the time of my visit, Studio 2's recording area still housed the remains of Abbey Road's highly successful "Beatles Exhibition," which during the course of two weeks last summer attracted some 17,500 visitors to this unique collection of Fab-Four Memorabilia. The studio had been laid out in a realistic recreation of a Beatles' session of the mid-Sixties, including band instruments, and the actual Studer J37 four-track used on those dates. A four-channel surround-sound mix made from original multitrack tapes had been prepared by Alan Parsons (who served as second engineer on several of the Beatles' latter sessions, including *Abbey Road*), with an accompanying video presentation of the band working in the studio, much of which, Vince reveals, had never been seen by the general public. The studio had hoped to tour the exhibition, but copyright clearances on some of the the film and video material has prevented this happening. Given the large huge crowds that managed to get to see the exhibition last year — Vince reports that additional presentations had to be added to the daily program at the last minute to accommodate the unexpectedly large number of people that came from all over the world to catch the show — Abbey Road was considering the possibility of staging it on a regular basis.

Moving on to Studio 1, the largest in the complex, I discovered that besides its normal fare of classical and orchestral sessions for both EMI and outside record companies, the studio is seeing an increase in film-scoring work. The room, with its extremely spacious floor area and tall ceiling, is capable of easily hosting a 120-piece orchestra, or a 250-piece choir. To enable retakes or overdubs to capture the same orchestral layout and acoustic feel as original takes, the studio floor carpet is arranged in numbered squares. Later this year Studio 1's small control room is scheduled to be relocated and increased in size, prior to the installation of a larger, more flexible console.

The Penthouse Studio was originally designed, Vince recalls, to serve as a self-contained facility for bands that wanted to set up for lock-out sessions, and avoid the inevitable crush of people that come and go in a studio complex the size of Abbey Road. The choice of a Neve 8108 console for the control room apparently was made so that visiting engineers wouldn't need to spend too much time familiarizing themselves with the recording hardware. Now, in fact, the room serves as a dedicated remix room, with a relatively small accompanying overdub and vocal booth.

Other rooms around the complex include Studio 3, which houses a wrap-around 36-input EMI/Neve console with NECAM servo-fader automation, a Studer A800 24-track, and UREI 813 monitors, plus four disk-cutting suites. Abbey Road also operates three mobile recording vehicles based on a Mercedes three-ton chassis, and intended mainly for remote classical sessions. Each van is designed to accommodate a variety of transportable EMI consoles, which can be built up from modular five-input subunits, and mated with a 16-track Studer A80 that also has been broken down into two separate packages — transport and electronics — for ease of removal from the vehicle, and subsequent setup at the recording venue; alternatively, the gear can be fitted in flightcases and air-shipped to anywhere in the world.

Townhouse Studios, located in Shepherd's Bush, West London, has an equally interesting pedigree. Owned by Richard Branson's Virgin Records conglomerate, and opened in 1978, the facility was intended to provide an alternative to the company's residential, out-of-town Manor Studio. (And while few bands and producers would object to a couple of weeks/months working in a gorgeous 18th Century manor house equipped with a state-of-the-art recording studio — and at £1,000/$1,500 per day, including accommodation for up to eight band members and crew, the Manor would appear to offer remarkable value for money — sometimes the pressures of big-city living conspire to prevent us from taking off for the green and pleasant Oxfordshire countryside; in such circumstances, an in-town facility fits the bill.)

The complex houses two studios and a pair of cutting rooms. During my visit, I discovered that Townhouse was planning a couple of interesting equipment upgrades, as well as buying The Who's old studio, Ramport, located in nearby Battersea, and which has been renamed "Townhouse Three." According to chief engineer Alan Douglas, the custom-designed, 40-input/32-group wrap-around

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THE FEATURES

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For additional information circle #59

can record at Bee Jay while our company shoots video. Supplemental shots of the artist will be made outside the studio environment and worked into the video production." **Eric Schabacker,** president of Bee Jay says, "Doing sound tracks for video is a logical progression for our business. We have already done soundtracks for **Judas Priest, Molly Hatchet, Axe** and **Krokus** that appear on MTV and other music oriented channels." *Orlando, FL.*

□ **VIDEOWORKS** (New York City) has completed a new computerized film-to-tape color correction suite. The new room is third in a series built by East Coast Systems for VideoWorks, and houses the new Bosch FDL-60 CCD Telecine attached to a Corporate Communications Color corrector for frame-by-frame, or scene-to-scene color correction. In addition to the new film chain, two Sony BVH-2000 one-inch videotape machines, CDL-480 Video Switcher, cassette duplication VTRs, and extensive audio and video monitoring gear were added. *24 West 40th Street, New York City, NY.*

□ **RBY RECORDING STUDIO** (Southbury, Connecticut) now offers half-inch video editing with special effects and character generation to better serve its clients interested in video. **Peck Brothers Band, Joe Walsh,** and **Dave Quinn** and **Ralph Jackson** completed recording dates and augmented their sessions with video shoots done on location, in the studio and at Toads Place, Waterbury, with the new equipment. *Rd 1 Main Street, Southbury, CT 06488. (203) 264-3666.*

□ **NIMBUS NINE RECORDING** (New York City) has installed a BTX SMPTE synchronizer for interlocking its MCI JH-24 multitrack to any other audio or video equipment. Staff engineers **Christopher Howard** and **Frank Roszak** completed mixing of additional music for the soundtrack of the no-nuke film, "In Our Hands," starring James Taylor and Pete Seeger. Film co-producer **Stan Warnow** supervised the mixing sessions for the June 12th Film Group. *1995 Broadway, New York City, NY 10023. (212) 496-7771.*

Western Activity:

□ **BONNEVILLE PRODUCTIONS** (Salt Lake City) has taken deliver on a new MCI/Sony Audio Layback machine that studio manager, **Dave Michelsen,** claims is the first of its kind in the Utah market. As Michelsen explains, "The layback machine is an audio-only recorder/reproducer fitted with heads that conform to the audio track configuration of C format one-inch videotape. This allows audio to be taken from a video master tape to a multitrack audio machine, enhanced or sweetened, and then to be layed back onto the video master tape. The advantages of this process are: 1) The machine can be maintained to a higher audio specification than is normally available in the audio section of a regular video recorder; this means a much higher quality audio track on the finished product. 2) Audio layback can now be done in an audio studio instead of a higher-rate video editing bay; this savings can be significant. 3) It allows an audio production house to keep control of the quality as all the audio work including layback onto the master video is done by audio specialists under one roof." Michelsen sums up: "The customer saves money and gets a superior end product." *130 Social Hall Avenue, Salt Lake City, UT 84111. (801) 524-2400.*

□ **EFX SYSTEMS** (Burbank, California) has taken delivery of what is said to be the first BTX Softouch Intelligent Audio Editing Controller, to supplement its Shadow SMPTE synchronizers. Fully programmable to suit any scoring, looping, mixing or sweetening application, the Softouch allows the synchronizer to be custom tailored to the needs of any project. The computerized system is capable of full function control of up to four slave transports via timecode, including record in and out, loop parameters, effects placement, and other synchronizable events. EFX is using the Softouch to facilitate its audio recording and film/video post-production services. These include 48-plus-track recording and video sweetening, layback to one-inch video on the new MCI/Sony Audio Layback transport, film dubbing, scoring, ADR, Foley and sound effects work. *919 N. Victory Blvd., Burbank, CA 91502. (213) 843-4762.*

Foreign Activity:

□ **STUDIOS 301/EMI** (Sydney, Australia) has taken delivery of an Audio Kinetics Q.Lock 3.10 syncronization system that can link multitrack audio tape machines to any video tape machine, and is being used extensively for feature film soundtracks, music for television commercials, and other video music projects, including FM/TV simulcasts. *301 Castlereagh Street, Sydney, N.S.W., Australia, 2000. (02) 20912.*

□ **TOYO RECORDING COMPANY, LTD.** (Tokyo, Japan) has installed a complete Ampex post-production system valued at more than $1.1 million in its new editing suite. The suite features five Ampex VPR-3 one-inch helical scan videotape recorders, a two-channel Ampex Digital Optics (ADO) system, ACE computerized editing system, AVC-33 production switcher, and ATR-102/104 and MM-1200 audio recorders. Designed to meet requirements for video production and videodisk manufacturing, president **Akira Sato** says, "We purchased the Ampex equipment as a system, rather than stand-alone products, because we believe Ampex's innovative power can be most effectively demonstrated in the total system. We believe the Ampex-equipped editing bay is the best in the industry, and we are demonstrating the results through actual operations." *Tokyo, Japan.*

ON THE STUDIO TRAIL

Mel Lambert at Large in London's Cosmopolitan Recording Community, visiting EMI-ABBEY ROAD, TOWNHOUSE, LANSDOWNE, AND GATEWAY STUDIOS

According to operations manager Peter Vince, EMI-Abbey Road Studios, now in its 53rd year of operation, has been going through a series of equipment upgrades during the recent few months, and also has additional plans for the spring of this year. Housed in a large three-storey Victorian mansion located in picturesque St. Johns Wood, North London, the complex certainly is impressive, and arguably one of the largest — if not the oldest — of England's studios; the facility even includes its own in-house, fully-stocked bar and canteen.

Kicking off my tour in Studio 2, Vince pointed out that the new 48-input/32-group Solid State Logic 4000E board currently gracing the control room is, in fact, the 100th Series 4000E console that SSL has produced. (A second 4000E is destined for installation in late April of this year; at the time of my visit a final decision of whether the board would end up in Studio 1 or 3 had yet to be reached.) As I discovered, the new SSL is the first off-the-shelf desk to be installed at Abbey Road. In the past the studio had either designed its own consoles, and then had them manufactured by Neve Electronics — hence the EMI/Neve designation — or ordered relatively "standard" Neve desks, and then modified them to its own requirements.

"We discovered though," recalls Vince, "that all the features we needed were standard on the SSL. For example, each channel [strip] has a built-in high-quality compressor [combined with an expander/limiter] which is a feature that had been specified on the early EMI desks." Other control-room equipment includes a 24-track Studer A80 (all Abbey Road studios are equipped with Studer

STUDIO — FACILITIES EQUIPMENT PEOPLE — UPDATE

Southern California:

☐ **MUZIC TRAK STUDIO** (Los Angeles) has upgraded from 4- to 16-track with the addition of a Studer A80 16-track tape machine. Additionally, the studio has acquired a Studer/ReVox A77 half-track tape machine, and an assortment of new microphones and outboard gear. Chief engineering duties are shared by **Randy** and **Courtney Branch**, with assistance from **Terrell Branch** and **James Johnson.** *2227 Alsace Avenue, Los Angeles, CA 90016. (213) 931-7508*

☐ **HIT SINGLE RECORDING** (San Diego) has replaced its Stephens 811D 16-track tape machine with a similar model Stephens that offers full transport remote control and one-inch tape capabilities, enabling the facility to now offer one-inch 8-track, in addition to half-inch 8-track recording and mixing. *College Grove Center, Lower Court 4, San Diego, CA 92115. (619) 265-0524.*

☐ **SUNSET SOUND** (Hollywood) has fitted a Neve NECAM I servo-fader automation system to its custom console in Studio 2. According to studio manager **Craig Hubler**, the recent upgrade represents the first installation of NECAM in a non-proprietary board. *6650 Sunset Boulevard, Hollywood, CA 90028. (213) 469-1186.*

☐ **A&M RECORDING STUDIOS** (Hollywood) has announced the appointment of **Don Hahn** to the position of vice-president and general manager of the studios. Hahn joined A&M in 1977 as senior engineer and has most recently served as director of operations. He is a member of the board of governors of the Los Angeles chapter of NARAS. *1416 North La Brea Avenue, Hollywood, CA 90028. (213) 469-2411.*

☐ **MIX MAGIC** (Hollywood) has opened a new post-production studio catering to the film and video industry. According to studio manager **Ruth Corbett** and chief engineer **Jim Corbett**, they used to make use of Motown Hitsville, where the company worked on Michael Jackson's "Beat It" video, and Lionel Ritchie's new video. Everything Audio's **Brian Cornfield** designed and equipped the new studio, which features a 40-input/24-bus Trident TSM console, UREI Time-Align® 813 monitors, Magna-Tek four-channel dubbers, JVC 6250 U-Matic videocassette decks, and an RCA telecine chain; Otari MTR-90 24-track and MTR-10 four-track machine, and Sony one-inch VTRs are also under consideration. *839 Highland, Hollywood, CA 90028. (213) 466-24420.*

— SOUTHERN CALIFORNIA PERSONAL-USE STUDIO SPOTLIGHT —

☐ **GINO VANELLI** and his brother Joe recently acquired all the components for their new home studio, based in Westlake Village, where they've just finished the first album on which the Vanellis have handled all recording and production duties entirely by themselves. The recent equipment purchases include an Otari MTR-90 24-track, an Otari MTR-12 half-inch, two-track, and a Lexicon 224X digital reverb. Everything Audio supplied the hardware package, and will be assisting the brothers in the design and construction of a second studio.

☐ **TITO JACKSON**, who currently is gearing up for the 1984 Jacksons Tour, recently purchased an Otari MTR-90 24-track to use along with his much-sought-after vintage API console, which will be installed as soon as construction on his custom-designed, Encino studio is complete.

☐ **MICHAEL SEMBELLO**, who is currently working on a follow-up to his hit "Maniac," has updated his personal studio in Pacoima with the purchase of an Otari MTR-90 24-track. This recent upgrade is said to be the first step in a complete renovation of Sembello's studio, where he plans to work on all his future projects.

☐ **UNITED WESTERN** (Hollywood) has taken delivery of a Mitsubishi X-800 digital 32-track machine, which will augment the studio's present X-80 portable two-track. "Our first consideration in selecting the X-800," says VP/general manager **Jerry Barnes**, "was the track configuration — 24 tracks just isn't enough these days; we had to go for 32. To me, the sound of the Mitsubishi is excellent." Barnes has already used the X-800 on an international session tracked in both London and Los Angeles. Having recorded a 60-piece orchestra, The Concertante Ensemble of London, onto 16 digital tracks at EMI-Abbey Road Studios, the tape was then hand carried to United Western, where a 200-voice chorus from Azusa Pacific University was overdubbed. "Machine-to-machine compatibility presented *no* problems whatsoever," Barnes comments. The resultant X-80 two-track mix will be used by Denon to master directly to Compact Disc for commercial release. *6000 Sunset Boulevard, Hollywood, CA 90028. (213) 469-3983.*

Northern California:

☐ **TRANSPARENT RECORDINGS** (San Francisco) has acquired a Nagra T-Audio transportable recorder, claimed to be the first in Northern California. The ¼-inch, 30 IPS machine will be used with a Studer 10/2 console for remote recording; it is equipped with center-track SMPTE for film and video production, and will also be available for record mastering and studio mixdown. *883 Golden Gate Avenue, San Francisco, CA 94102. (415) 563-6164.*

☐ **PATCHBAY STUDIOS** (San Rafael) has recently added a Sony PCM-F1 digital audio processor, and a Decillionix DX-1 digital effects program for its Apple computer, enabling the studio to offer digital mastering, input sampling, a variety of digital effects, and remote digital 2-track recording. *2111 Francisco Blvd., San Rafael, CA 94901.*

☐ **LIVE OAKS RECORDING** (Berkeley) was recently completed and features an MCI-equipped 24-track control room with a large selection of signal processing equipment, including a Lexicon 224X digital reverberation unit. The studio was designed by **Sonic Landscapes** for use primarily as an overdub room, and offers access to a wide selection of electronic keyboard instruments. *1300 Arch #2, Berkeley, CA 94708. (415) 540-0177.*

☐ **THE PLANT STUDIOS** (Sausalito) has been sold by its previous owner, Laurie Necochea to **Stan Jacox**. According to general manager **Paul Broucek**, immediate plans following the sale call for a cosmetic facelift of the three-room complex, including increasing the size of Studio B's recording area, and also altering the acoustic treatment to enhance the room's liveliness. Present studio staff remain as before, Broucek adds, including **Mick Higgins** as chief of maintenance. The studio plans to hold a grand re-opening party in the late Spring. *2200 Bridgeway, Sausalito, CA 94965. (415) 332-6100.*

— AUDIO/VIDEO UPDATE —

Eastern Activity:

☐ At **CIANI/MUSICA** (New York City) an ENG production team from the TV Arts School of The Center For The Media Arts, New York, recently set up shop to shoot a recording session of a GE "Beep" Dishwasher commercial. The edited 10-minute videocassette of the session was incorporated by **Suzanne Ciani** (seen right) as a learning tool during the first Technology, Entertainment and Design Conference held late February in Monterey, CA. Ciani is a New York-based composer, arranger, producer, and performer who has created commercial electronic music for a wide variety of advertisers. The CMA team was directed by **Julio Cintron**, with **Fernando Medina** and **Ole Shapiro** serving as cameramen, and **Mark Gainor** editing. *New York, NY.*

CIANI/MUSICA — video/jingle session

☐ **AMERICAN BROADCASTING CORPORATION** (New York City) has had Ampex ship 12 VPR-3 helical scan videotape recorders to Sarajevo for the ABC Television Network to use in its coverage of the Winter Olympic Games. ABC signed a $10 million contract for 100 VPR-3 recorders for coverage of the Winter and Summer Games and political conventions. Following the games, the video recorders will be used to update and expand ABC's tape facilities. *New York City, NY*

☐ **VISUAL IMPACT** (Orlando, Florida) recently formed an agreement with Bee Jay Recording Studios, whereby the two firms will co-produce video demos on behalf of musical acts. According to **Craig Soldinger**, vice president of Visual Impact, "We have had tremendous interest from artists who want videos of themselves for various purposes. [We] have reached an agreement where the artists

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STUDIO UPDATE

FACILITIES · EQUIPMENT · PEOPLE

North East:

□ **BEARSVILLE STUDIOS** (Bearsville, New York) has acquired an AMS DMX15-80S stereo digital delay, which has two channels of delay (1.638 seconds and 409 milliseconds), two channels of pitch change with de-glitch cards, and digital loop editing. In late winter '84 the studio will be replacing the Neve 8058 currently in use with a custom Neve in-line, 40-input 8088 fitted with NECAM II automation. *P.O. Box 135, Speare Road, Bearsville, NY 12409. (914) 679-8900.*

SKYLINE — New Outboard Effects

□ **TROD NOSSEL RECORDING STUDIOS** (Wallingford, Connecticut) has upgraded its monitor system with the installation of JBL 4430 bi-radial monitors. Lang PEQ-1 equalization was also added to the studio. *Wallingford, CT 06492. (203) 269-4465.*

□ **SKYLINE STUDIO** (New York City) has added a Lexicon 224X digital reverb, a Marshall Time Modulator and Tape Eliminator, a rack of API Equalizers and a Delta Lab Effectron II to its sound processing equipment. In addition, a restored 1896 Steinway B grand piano, an Oberheim OB8 synthesizer, LinnDrums, a Roland Jazz Chorus amp, a custom built MacIntosh/Alembic bass amplifier, a set of Yamaha Recording Series drums, and Zildjian cymbals have been added to available musical instruments and equipment. *36 West 37th Street, New York, NY 10018. (212) 594-7484.*

□ **CRYSTAL CITY TAPE DUPLICATORS** (Huntington, New York) has installed two new Otari master recorders to complement its Otari DP7500 bin-loop duplication system: an MTR-12 ¼-inch two-track and an MTR-10 half-inch four-track. Other recent additions include Dolby Model 361 noise reduction units; a pair of JBL 4411 reference monitors; a new cassette packaging system; and an Otari Model DP1610 quality control monitor. *48 Stewart Avenue, Huntington, NY 11743. (516) 421-0222.*

South East:

□ **EDDY OFFORD STUDIOS** (Atlanta, Georgia) has recently added a new Studer A800 24-track tape machine. Other new equipment acquisitions include an NED Synclavier, Simmons drums, and a Lexicon 224X digital reverberation unit. *P.O. Box 90903, Atlanta, GA 30364. (404) 766-5143.*

□ **BEE JAY RECORDING STUDIOS** (Orlando, Florida) announces the resignation of chief engineer **Bill Vermillion**, who has been with the studio for 16 years, to "pursue one of several private interests." Studio president **Eric Schabacker**, says Vermillion was responsible for a large part of the design of Sphere's Eclipse C console, the prototype version of which is in operation at Bee Jay, and was very instrumental in the design and construction of the multi-studio facility. Bill's track record also includes Platinum and Gold albums for engineering. "Bill was an integral part of the Bee Jay family and will be missed by the entire staff, not only as a co-worker but as a friend," adds Schabacker. *5000 Eggleston Avenue, Orlando, FL 32810. (305) 293-1781.*

□ **REFLECTION SOUND STUDIOS** (Charlotte, North Carolina) is installing a new MCI JH-636 automated console in Studio A. The console features plasma metering, parametric EQ, and Spectra-Vue. JH-24 24- and half-inch two-track recorders and Lexicon 224X digital reverb with LARC option have also been added to Studio A. Sessions are already running in Studio C with the room's new MCI JH-636 board, featuring VCA fader grouping and interfacing with a new JH-24 recorder. dbx 900 racks with limiters, gates and de-essers have been added to both rooms. *1018 Central Avenue, Charlotte, NC 28204. (704) 377-4596.*

□ **STRAWBERRY JAMM STUDIOS** (West Columbia, South Carolina) have added an MCI JH-114 multitrack with Autolocator II, Ampex AG-440B two-track recorder with remote, three UREI 1176 limiter/compressors, Sansui and JVC cassette decks, and new JBL 4435 dual-15 bi-radial main monitors powered by Hafler DH-200 Class A amplifiers. These additions complement the existing Neotek Series III console. *3694 Apian Way, West Columbia, SC 29169. (803) 356-4540.*

South Central:

□ **DISC MASTERING INC.** (Nashville) has purchased a Neumann SAL-74-B cutting amplifier. Due to the use of an electronically balanced circuit replacing the transformers once used a the modulation and feedback inputs, the SAL-74-B is said to offer improved phase response and greater feedback at low frequencies, leading to an increase in channel separations at those frequencies. "To the best of my knowledge, this amplifier is the only factory-built Neumann of its kind in Nashville," stated **Randy Kling**, president and chief engineer of DMI. *Thirty Music Square West, Nashville, TN*

□ **DALLAS SOUND LAB** (Las Colinas, Texas) recently hosted a grand opening to demonstrate the new three-studio audio facility. Studio A is a 48-track room set up to record up to 60 players for multitrack sweetening to picture. Studio B is a 24-track video interlock room for voice-overs, dialog replacement and video sweetening. Studio C is a film mixing and screening theater offering a four-channel Dolby sound system similar to Lucasfilm's THX design, a Foley effects area, a permanent 22-foot screen, and has microprocessor-controlled projectors (16- and 35mm) and dubbers. In addition, Dallas Sound Lab is hard-wired into the three sound stages some 250 feet away in the main building of the Dallas Communications Complex, which enables groups to do live music shooting, in video or film, with synchronized multitrack sound. *Four Dallas Communications Complex, Suite 119, 6305 North O'Connor Blvd, Irving, TX 75039. (214) 869-1122.*

□ **EDENWOOD RECORDING STUDIO** (Dallas, Texas) has taken delivery of a Studer A800 24-track tape machine. "The Studer adds to Edenwood's audio for video sweetening capabilities, which now include a three-machine lockup via BTX synchronizers and on-screen display of timecode by BTX's Cypher SMPTE reader/generator," says owner **Jerry Swafford**. Other recent additions include a Studer phone patch system, and a new alliance with Frame by Frame Associates, a facility for stills, film, video and stock photography, headed by **Gary Campbell**. The new association between the two companies was formed to better serve film and video clients. *7319-C Hines Place, Suite 201, Dallas, TX 75235. (214) 630-6196.*

EDENWOOD — New Studer A800

□ **COOK SOUND** (Fort Payne, Alabama) has recently upgraded to its microphone and instrument collections with the addition of Neumann U-48 and U47, and Sony ECM50 microphones; Oberheim DMX drum machine; Yamaha 7'4" grand piano; and a Hammond C-3 organ. Other recent additions to the 24-track Studer/Neve facility include Yamaha NS-10M monitor speakers, MacIntosh 2300 power amps, and an URSA Major reverberation unit. *P.O. Box 67, 1419 Scenic Road, Fort Payne, AL 35967. (205) 845-2286.*

Mid West:

□ **CHICAGO AV** (Chicago, Illinois) has just undergone several hundred hours of rewiring and special studio maintenance while the SuperSpots facility was shut down for vacation. Oberheim OB-8 synthesizer and DX drum machine were added to the instrument list. **Lori Marie Cerny** has joined the staff as director of administration and finance. *216 West Ohio Street, Chicago, IL 60610. (312) 280-9433.*

THE TACT FACTOR

pants "down off the ledge" is to *overtly* (and loudly) ignore the possibility that the hostilities could escalate any further. Invite them to continue their discussion (and make sure you emphasize talking rather than smacking) in the studio lounge, waiting room, hall, or what have you. The change of physical location alone will provide a cooling off period, and will protect the gear. And make sure no spectators are allowed in the new "arena"; this spares everyone their pride, thus encouraging professionalism. Let 'em work it out on their own — and don't rush them. The clock is running, studio is making money, and passion is a goodly part of musical product anyway. In short:

Tact Factor Axiom #3

People will usually behave rationally if they can be shown that it's in their best interest to do so — in a manner which does not insult or publicly embarrass them.

With that in mind, let's look at a few people and situations along the Ol' Session Trail:

Teddy Triple-Scale knows the only way to get his special sound. And he'll tell you just how to do it, despite the fact that it might depend on factors (technical, artistic, emotional) that he's never considered. Then again, *he might be right*. Don't let your pride get in the way of learning something new. The best bet for a Best Possible Track: set it up his way; set it up your way; let the Boss decide. Or, print both (if you can afford the tracks) and reserve the decision for a time when there's no emotional heat on the issue.

Veronica Vocalist is personal friends with the producer. Mr./Mrs. Producer may not be able to direct (i.e., criticize) Veronica. Try to establish with the producer if they would find it easier for the comments regarding the best road for a re-take to come from your mouth to Veronica's cans. Their relationship may be fine in other capacities but, in the studio, you may have to be the go-between (lucky you).

Nick Nervous-Novice gets so caught up in the details of what he thinks should be happening, that he forgets to use his own ears. He'll tell you exactly what you're doing wrong, and exactly what you should be doing to correct it. He's read a lot about what a recording studio can do, and he's sure that you can fix the out of tune concert B-sharp on his trumpet track by adding a little flanging right at that spot. (Well, can't we just put the guitar through the Harmonizer?") Resign yourself to the fact that he'll never be satisfied, and try to calm him out by involving him in as many decisions as you can. (If he complains later, you can politely suggest that he was in on the bad decision.) If he asks for more compression during a power guitar overdub, and you're already showing a lot of gain reduction on a completely clipped out amp, suggest to him that you can always tighten up the track as you mix, but that it might be "smarter not to get locked in now." If he insists, give it to him, and tell yourself

"Then again, he might be right. Don't let your pride get in the way of learning something new."

that it won't make or break a hit tune anyway (which is true).

Murray the Studio Manager's concept of Studio Policy is to wait till something goes wrong, then yell a lot about it when it does. You can never seem to get a clear line on what should go down in any particular situation. Murray may have a need-to-exert-authority problem. It's rather like a drinking problem; the poor wretch may need his fix. Your best course is to dis-involve yourself with the emotional side, and play it strictly on its rational merits. In other words, don't get into it with him. Not in front of a client, not behind closed doors . . . not at all.

Light-gauge Lefkowitz and Dobro Danny are guitar players in the same band. Part of what makes the band so good when they play is the dualing leads they get into. Meanwhile, back in the control room, Danny wants you to tell him that you've recorded a lot of guitar pickers in your time, and that you think he's got the best sound around — or at least it's better than Light-gauge's. Mr. Lefkowitz needs to hear that the special distortion box his brother made for him gives him a hands-down edge over Danny's stock music store amp.

Well, you know what to do. Shift the emphasis back to how well they interweave with each other — if, that is, you have to say anything at all. Avoid the temptation of giving different stories to different people just to preserve the peace. The ultimate Tact Factor master stroke keeps everyone in touch with a single reality, without threatening anyone's position.

Thus we see how the brave Lone Engineer can come ridin' into the Control Room astride his trusty technotactful steed and dispense with the bandits. The real trick is to avoid becoming a bad guy yourself. ■■■

. . . to be continued —

THE TACT FACTOR

highly accurate. He deserves the "Tact Factor Award" and, much more importantly the respect of his fellow professionals.

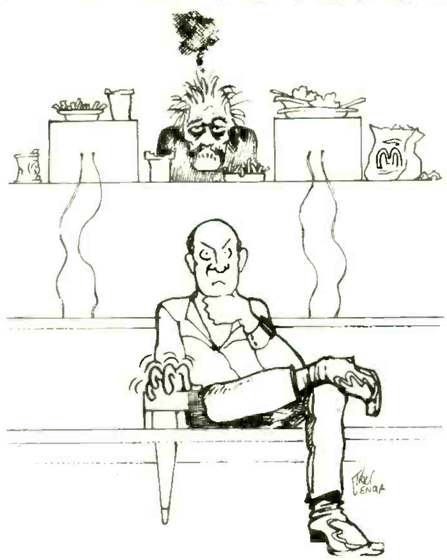
Above all, if you possibly can, it's a good idea to be somewhere else when the money changes hands. When it comes time for the musicians to be paid, try and find something to be busy with. (My favorite dodge is to either join the break-down gang in dealing with mikes and cables, or to be fussing with levels on rough mix copies). This accomplishes two things. Not only does it leave the parties involved freedom to dicker, it also dis-involves you and the studio from any potential bad-blood resulting from the all-too-frequent last minute wheeling and dealing. ("But on the phone you said: 'Triple scale, and a coupon!'.")

If the client is paying the studio at the time of the session, try to have as little involvement in this interaction as you can. Don't even be in the room unless one or both parties want you there. If you're on the studio payroll, all that should ever be required of you is an accurate account of time and materials used, and/or a straightforward technical description of downtime.

If you're an independent you must make it clear to whatever producer and/or artist you are working for that *you will not lie for them*. Try to establish this as early on as possible in the relationship.

Fortunately for you, technical debates are generally easy to resolve. Either the studio works, or it doesn't; you'll know it (or you should) and test gear will show it.

For whatever it's worth, I don't approve of engineers taking money from the studio "under the table." A legitimate commission or finders' fee is one thing; a silent kickback that the artist and/or producer isn't supposed to know about can only become an obstacle to the Best Possible Track. It puts a psycho-defensive wall between engineer and artist, precluding sincerity and eye contact, and many of the other subtle tools of the Tact Factor. If we engineers do nothing else, at least we shouldn't get in the way of the music; our equipment usually does enough getting in the way as it is.



"We've all had to deal with the Overbearing Producer, for whom nothing seems to be good enough."

Say What?

Knowing what to say is only slightly less important than knowing what *not* to say. A particularly tough time is drawn by the Maintenance Person called into the studio for a quick fix. While most of the left hemisphere of the brain is busy with a point-to-point analysis of what might be troubling the ailing gear, most of the right hemisphere must feel out the "vibes" in the room, and try to cool out everyones' anger/disappointment. And the Maintenance Person has to do all of this, usually without much background as to who is who and what they might be feeling. Truly it's the case of a doctor needing to develop an instantaneous, situation-specific bedside manner. The "patient" is a mechanical or electronic device, but the "relatives" are human.

Sometimes things get out of hand. Pete the Producer less than politely suggests that Bob the Bass Player may have a limited understanding of how and when to put his fingers where. Bob responds with some speculation concerning Pete's choice of possible companions, probable progenitors, and preferred methodology of personal recreation. The best thing to do under these circumstances is to batten down the hatches and wait out the storm. Keep your mouth shut and your ears open (your grandchildren may crave the eyewitness details when they take computer-assisted Pop/Rock History 101 someday), and look for the most delicate piece of equipment to protect from flying missiles and bodies. The console is a good bet.

I knew an engineer/studio owner who kept an official Little League "Louisville Slugger" baseball bat (same length — greater mass than a law enforcement agent's nightstick) concealed in the space between his 16- and 24-track, about 10 inches from his left hand when operating the first bucket of console faders. He called it, fondly, "The Equalizer." Being a man of superior tact, judgement and no small mass himself, he used it merely to threaten (a true story).

Seriously folks, I've been in on more than one session where physical violence became a real factor; not fisticuffs and bloodletting, mind you, but there was a little pushing and shoving. The best way of getting the particifairlight

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THE TACT FACTOR

to the clarity of delineation of the pecking order in that situation; and (to lesser extent) the experience level of the parties involved.

The (Wo)Man in the Middle

The brave engineer's role ain't always so easy. You have to answer to *everyone*: the studio, the producer, the artist(s), the money people, the Company people, the promotional people, the hang-arounds, the hangers-on, the great, the near-great and the think-they're-great; your own business/professional sense, your own creative head and, above all, your own conscience. Not to mention your lover ("Sorry babe, looks like the session's gonna run long . . . again").

There's such a difference between the way an instrument sounds to the ear and to a microphone/loudspeaker — even without all our toys — that engineers generally are regarded as practitioners of (relatively benign) black magic. As such, we're called upon to explain the whys and wherefores, and alleviate, circumvent, and otherwise nullify the fears of the uninitiated. (Sounds like The Wizard of Oz, no?) And, to make matters worse, we usually have to mediate those lovable, laughable Control-Room-Differences-Of-Opinion — and, while doing so, manage to save the honor of all the above mentioned individuals. Tall order.

If you are not the producer (but just the engineer) and/or the Boss is present, your main mediation job is to *present the range of options* in any given situation: "I can make the vocals 'fat' and 'churchy' with this mike and EQ, or 'light' and 'floating' with that combination." You don't have to stick to technical terms, but you must know exactly where you stand before venturing too far into the great Gray Area of musical suggestions. Usually, there are too many conflicting creative viewpoints on any particular session as it is; we should be careful not to add any of our "noise" to the signal path.

"Watch closely when someone makes a mildly controversial statement . . . the focus of that fraction-of-a-second dart of the eyes is your boss."

Engineers need to think in three languages: the Language of Physics, the Language of Music, and the private dialect of the Native language that all sessions (at least the extended ones) seem to develop. (Some bands definitely speak in their own code, right?) The design and construction of your console is precise; the "design" of show-biz talk is *not*. Words mean subtly different things to each of us ("I hear ya . . . "), and every word has different personal or mental pictures associated with it. They will want you as their engineer if you "speak their language" or, in the ideal case, if you wear their ears as well as your own. We all know that "bad" means something entirely different to an R&B producer than it does to the AOR man from the serious music division, but what does "dark" mean, and to whom? Where do you stop equalizing for "edge," and start EQ'ing for "air"? "Pitch" translates fairly as "frequency"; "tone color" as "timbre." But what about "punch"? Thus:

Tact Factor Axiom #2

Your success in a working relationship is governed by your ability to understand and "feel" from everyone else's position as if it were your own.

Sooner or later (usually sooner) somebody is going to ask you for a musical judgement that you really don't want to make. If you have excellent control of your voice, and the sincerity of a con man, you can get away with answering the question: "Well, what do you think?" with such non-polemical responses as: "Powerful!"; "A lot of feeling there . . . "; or "I don't honestly think we can get better than that."

Most of us are not so gifted or so quick to put our hard-won reputation for honesty at risk. Thus, we're left with the necessity to duck the issue entirely. If we elect government officials to be professionally good at side-stepping, why can't we do it ourselves (in the line of duty, of course)? The best method is to hand it back to them: "Well, why don't we hold that one and do one more. And you guys can compare them and decide."

This response is not only good for the tape manufacturers, it's good for the studio; although "running the clock" is an underhanded way to make a buck, and runs the risk of client alienation. (It's best to have a *real* good reason for any time-consuming procedure that we do — unless we're working a spec deal, or something like that). I'm a strong believer in encouraging the artist to make as many of the creative decrees as possible. It usually takes longer that way but the product hangs together better. And it goes a long way towards stopping squabbles in the studio. Any tension you can spare the performers is worth going for.

Say Too Much, Say Too Little

Sometimes there's absolutely *no* doubt about who the Boss is. We've all had to deal with the Overbearing Producer for whom nothing seems to be good enough, and no matter what you do, you'll be told that it's the wrong approach. (Usually because the Boss didn't think of it first.) You try to be helpful and always seem to end up getting shafted for it.

Intellectually, you know that he/she is the boss, and that they too have "got to serve somebody," but on the gut level you're still feeling: "Hey, who *is* this clown who doesn't know the room; doesn't know the equipment; doesn't know his/her track assignment from a whole note; has no apparent auditory abilities; and is telling me in front of everyone how to do my job?!"

And so your creative side pulls a sour-grapes act. You go on auto-pilot (minimum commitment status) and you get quiet and sullen and, somehow, you can't see your way clear to making helpful suggestions anymore.

This is dangerous, since everyone will consider your silence as either "spoiled-brat-syndrome" (if they know how the Boss has been treating you), or lack of knowledge/ competence (if they don't). A sympathetic and supportive second engineer can be a great help here. If you don't have one (and even if you do) try to do an unusually good job — just for spite. Keep smiling!

Some of us (like me) suffer the need to express our opinions too frequently. It's a good idea to keep a constant watch on your own emotional VU meters, and know your own motives for making any particular statement. "Am I really helping to get the Best Possible Track, or am I just honkin' my own horn?" Us bigmouths need to be ruthless in our self-monitoring. In other words, be sure your own head is in alignment before punching into any debate.

Maintaining a laid-back attitude has a hidden advantage. When you finally do say something (assuming it's not asinine) it'll carry more weight. There's a tour sound engineer for a major progressive rock band who has never been heard to say anything above a whisper. His quiet manner predisposes his associates to really "listen" to what he has to say; and what he says is always

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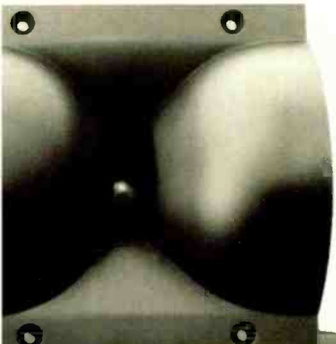

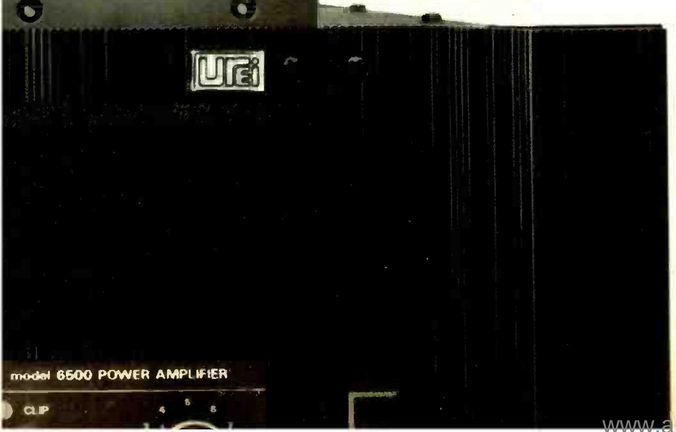

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SESSION DIPLOMACY

THE TACT FACTOR

Original illustrations by TRICI VENOLA

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

by David Brody

Okay, so you know all about the system gain structure, the recorders, and every toy in the outboard rack. You've got the console layout under your fingers, and can play it in your sleep. Using your extensive musical background, you make edits so smooth and tight that they all swear the chart was written that way . . . and you're intimate buddies with every microphone in the world. In short, you've got your technical chops up to an unbelievable level, Congratulations, Kimosabe, you've won half the battle.

I welcome you now to the dark realm of personalities and egos wherein intellect, rationality, and the stuff you learned in Engineer School are but "expression modules," totally dependent on the patching of stronger and deeper forces for their meaning flow. It's an emotional land where *what* you say is not as important as *how* you say it; an "unfair" universe in which the way your client perceives you is of vastly greater value than your technical qualifications.

Your most powerful tool in dealing with these matters can quiet things better than any kind of noise reduction, even out heavily transient outbursts less noticably than any compressor, and remove unpleasant feedback peaks more precisely than any filter set. It adapts itself to both balanced and unbalanced situations, but it's tough to get a handle on — and no two engineers use it the same way.

Let's call it the *TACT FACTOR*.

No matter what anyone tells you, only *you* can develop your own rap. Only you can discover and work out the rough spots in your professional personality. And only you know the particulars of your projects. But sometimes it's helpful to look at situations in abstract; "at arm's length," for the sake of perspective. Those differing with the views expressed herein are invited to write their own article.

You see, life is plenty rough out here on the Cultural Frontier. Some of the Artistic Injuns is "friendly," but some of 'em get a might ticked-off when you steal their Creative Hunting Grounds . . . and there's no shortage of Entertainment-Industry-Outlaws. So let's get our Tact-tactics together, podner; there's a whole passle o' problems to deal with.

Who's The Boss Around Here?

Right from the git-go, we're faced with the "How Much Creative Input Should I Express?" question. We all crave the ideal situation: Being given complete freedom to do the tracks any way we want. The recording game is no longer a question of getting the "best" (i.e., cleanest) sound from studio to disk (or soundtrack), but one of getting the most "appropriate" sound for the project. You must determine (a.s.a.p.!) who makes the final decision as to what that most appropriate sound is. Yeah, it's supposed to be the producer but, as we all know, there are producers and then there are producers (Definition: *Producer* — the one most likely to ask, "How many tracks we got left?") Engineers go for the Best Possible Track; the strongest artistic performance recorded with the most appropriate sound. Things are not always what they seem. The guy who's been introduced to you as the producer may not be the dominant person. Nor is the individual who talks the loudest (or longest). And it's not necessarily the money person. It might be the quiet lady sitting in the corner.

In just about every human situation there's a Boss. Gauging the boss's pleasure (or displeasure) with what's going down is the best indicator of how those involved will regard the final product and your performance. Learning who the Boss is can be the major part of psyching out the social scheme of things. There's a behavioral game that we humans play. I call it: "Everyone look up to the Boss" (aka: "The eyes have it . . .").

Watch real closely when someone makes a mildly controversial statement or raises a question of direction. Most of the people in the room will give a quick glance at one particular person. The focus of that fraction-of-a-second dart of the eyes is your Boss. (*Caution:* The person who first answers the question, or speaks to the statement, is likely to be the one who is most threatened by it, and may *not* be the Boss. And if not, the speaker will check out what he or she is saying with the same glance at the Boss.)

"And some have greatness thrust upon them." Here's the real Freaker. You may find that when a controversy comes up, they all look at you! An engineer behind his or her console can be a pretty impressive figure (that, as we all secretly know, is why we became engineers in the first place). You then have three options:

1. You can defer the privilege of answer to the one you figure to be the Boss and, without seeming disinterested, suggest to them that you don't want to get in the way of their creativity.

2. You can give an "information only" answer; i.e., play the roll of technician/recordist.

3. If you have high personal hopes for the future of the project, and are willing to take the gamble of your own ambition, you can express a creative opinion. If you're employed by the studio, you'd better make damn sure that everybody knows its *your* personal opinion. Humble is safest.

So, Let's Get Metaphysical and get-down-with-the-mellow sound of:

Tact Factor Axiom #1

The incidence of inter-personal hassles in any given situation is inversely proportional

— the author —

In his gainfully employed eight years, **Dave Brody** has engineered 37 albums (from the sublime to the ridiculous) more than 75 soundtracks for films and multimedia presentations, and "enough demos to keep the average A&R Department busy for a year." Known around the New York studio scene by his curiously appropriate initials (DB), Brody has worked both in television and as a musician/vocalist, which might account for his oddball perspective. Citing the Industry's crying need "to have even **more** people calling themselves Producers," Brody has, of late, been producing and co-producing several projects for commercial release.

— AUDIO/VIDEO MOBILE UPDATE —

☐ **GREENE, CROWE, AND CO.** (Los Angeles) recently recorded a two-night **Police** concert at the Omni in Atlanta, Georgia for a special video to air on Showtime. **Eddie Offord** engineered the 46-track date using two Otari MTR-90 and two Ampex MM-1200 24-track tape machines. Two of the machines were used in tandem, with the other two taking over as the first set ran out. An Auditronics mixing board with 50 inputs handled mixing duties. Assisting Offord was **Chuck Fedonczak**. The video shoot was directed by **Kevin Godley** and **Lol Creme**. *3083 N. Lima Street, Burbank, CA 91504. (213) 841-7821.*

☐ **AL TEARE RECORDING** (Pittsburgh, Pennsylvania) recently completed the upgrading of its remote van. New equipment is centered around an Auditronics 110-8 console and an Otari MX5050-8SD multitrack. Monitoring is handled by JBL 4311s, powered by a Crown D-150 amplifier and tuned using a Klark-Teknik DN30/30 dual graphic equalizer. New outboard equipment consists of a dbx 208 noise reduction system, two dbx 903 compressor/limiters, and two dbx 905 parametric equalizers. Two Electro-Voice RE-20 microphones as well as a pair of Countryman Associates Type 85 direct boxes have been added to the mobile's existing microphone collection from Neumann, AKG, Beyer, Shure, and Sony. *9076 Willoughby Road, Pittsburgh, PA 15237. (412) 367-1526.*

☐ **GBH PRODUCTIONS** (Boston, Massachusetts) recently completed three live recording projects for D.I.R. Broadcasting. On-location multitrack recordings of Canadian artists **The Payolas** and England's **Chaz Jankel** and **Nick Heyward** will be aired on the nationally distributed King Biscuit Flower Hour radio series. *125 Western Avenue, Boston, MA 02134. (617) 492-9273.*

☐ **OMEGA AUDIO** (Dallas, Texas) recently recorded a live album by **Big John Hall** for independent release in the Gospel field. The recording took place at Bethesda Community Church in Ft. Worth. Producing was **Dan Smith,** with engineering by **Randy Adams, Paul Christensen,** and **John Carey**. In addition, the Omega rig was hired by R.J. Productions in Houston to handle the remote audio recording for a new 13-week syndicated television series titled "Jimmy Dean's Country Beat" The series was shot in Houston at the Virginia City Club, where 28 acts were recorded over a two-week period. Audio engineering was by **Marvin Hlavenka, Michael Parks and John Carey**. *8036 Aviation Place, Box 71, Dallas, TX 75235. (214) 350-9066.*

☐ **MOBILE AUDIO** (Rome, Georgia) has recently updated its 45-foot mobile truck to include an Otari MTR-12-4-OB half-inch 4-track, in addition to its present MTR-90 24-track machine. Six new Sennheiser MD431 microphones have also been added to a large supply of on-board microphones. *National City Bank Bldg., 3rd Floor, P.O. Box 6115, Rome, GA 30161. (404) 232-7844.*

☐ **VIDEO GENERAL** (Long Beach, California) has just acquired a new 24-foot mobile video truck with 10-camera capability, machine rooms, and an isolated audio booth. The audio booth features a Yamaha 12-in console, and 8- and 2-track tape machines. In addition to a full line of mobile video equipment, including Ampex 1-inch C-type VTRs, CDL-850 switcher with double re-entry and Vital extended effects, and Sony ¾-inch U-Matic VCRs, the company can presently offer off-line editing and duplication facilities. *1200 E. 2nd Street, Long Beach, CA 90802. (213) 437-7569.*

☐ **MIDWEST VIDEO INDUSTRIES** (Kansas City, Missouri) has unveiled a new 40-foot video mobile which features five Philips Triax cameras, Sony BVH-1100A 1-inch video decks with slow motion controllers, a Grass Valley Group 1600-7K switcher, a 26×16 Auditronics audio console, and a Chyron 4100 character generator. MVI's association with Video Production Services (VPS) and Sound Recorders is said to offer location producers top-quality post-production capabilities. Both facilities, located in the same building in Kansas City, can be used simultaneously to edit raw footage. "A producer can edit his video footage with CMX and Quantel at VPS, while the audio is remixed at Sound Recorders," says **Bob Streeter**, VPS communications consultant. "Both facilities are connected physically and electronically." *3947 State Line Road, Kansas City, MO 64112. (816) 531-3822.*

ENACTRON MOBILE — *continued . . .*

a Yamaha PM1000 submixer, two Stephens 24-tracks, two half-track Scullys, Dolby noise reduction, Bryston Pro II power amps, and monitors by Klipsh, JBL, and Advent.

At the time Jones' staff contacted the Los Angeles-based office, the Enactron Truck was in Denver, where truck manager/engineer Stuart Taylor had just finished recording a live satellite broadcast of Lynn Anderson for the Nashville Network cable system. So, while Taylor readied the truck for the trip from Denver to Seattle, engineer Alan Vachon busied himself with preparations in Los Angeles.

Through conversations with the show's staff in L.A. and Seattle, it was determined that they would need an extra console to handle the 63 inputs required for the concert. As a result, a Yamaha PM2000 was rented for use in conjunction with the truck's Neve and Yamaha boards. Also needed were an additional 48 transformer/splitter boxes, and two wireless microphone transceivers. These, along with the necessary cables, supplies, etc. were loaded into the same plane that Taylor and Vachon took to Seattle on Thursday night.

On Wednesday evening, Taylor and Vachon had attended a rehearsal with Jones and his band, Patti Austin, James Ingram, Ray Charles, and a 17-piece horn section, at The Complex, in west L.A. The orchestra and choir were from the Seattle area, and would not be able to run through the entire show until scheduled rehearsals Friday night and Saturday afternoon. Nevertheless, the Enactron crew got a better idea of what they'd be facing in Seattle.

Load-in for all the lighting and PA equipment and the band's gear was eight o'clock Friday morning. During this time, Taylor and Vachon installed the Yamaha PM2000 in the truck's "comfort zone" — an area at the front of the truck used for overdubs, etc. — and wired a separate monitor speaker. The additional 48 splitters were combined with the truck's original 48, so that the mobile received a direct feed from the stage signals, and the PA system took its feed from the transformers.

It was early afternoon before the crew were able to run the cables. The 72 mike lines were run to the splitter boxes in three, 24-pair cables. The talk-back line ran to the monitor console, and a mono-audio and SMPTE timecode feed were run to the video company, which distributed the SMPTE code. The two wireless mike transmitters were set up at the back of the auditorium, and four other mikes interspersed through the center and front for ambience pick-up.

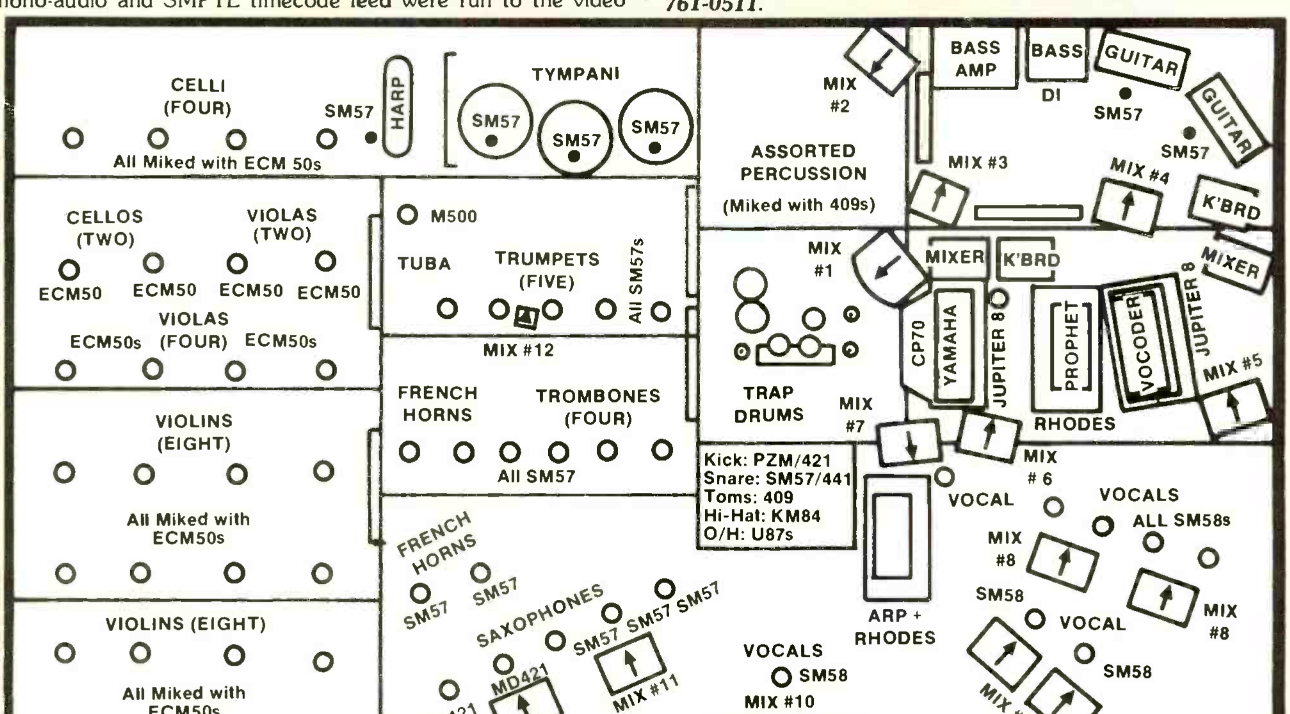
Distribution of all those inputs was handled by using the Yamaha PM2000 exclusively for a submix of the rhythm section — guitar, keyboards, drums, tympani, and bass — and directly patching the PM2000's outputs into eight tracks on the multitrack. Audience mikes were submixed on the PM1000, while all other inputs — harp, string and horn section, lead and background vocals, plus choir — were handled by the Neve console.

Of the total 84 mikes on-stage, some combining was done before the split. Specifically, 29 orchestra mikes were paralleled into nine outputs (two cello, two viola, four violin, and a harp), the 17 mikes in the horn section combined into eight (solo sax, lead sax, saxophone section, trombone section, french horns, lead trumpet, and trumpet section). The rhythm section received 17 mikes and 12 direct lines. A pair of Neumann U87s covered the 41-voice choir, plus four Shure SM-58s for the primary vocals, and four for background vocals.

After all the lines were checked, it was already evening and time for the first full rehearsal and sound check. Basically, the rehearsal was for the benefit of the band, orchestra, and choir, as well as to help Jones' engineer, Bruce Swedien, oversee the mixes in the truck, and the house sound. In the truck alone, Taylor and Vachon were handling three mixes —the multitrack, a stereo half-track mix, and the mono-feed for video. The rehearsals also provided the lighting and sound companies with an opportunity to become familiar with the show. The Enactron crew took the opportunity to evaluate individual instrument sounds and mike placements.

On Saturday night, the Seattle Celebration packed the house. Vachon attributed the show's success to the professionalism exhibited by all those involved in pulling it together. And, as is true with most people involved with the production of such an event, Enactron's Stuart Taylor and Alan Vachon commented that they had more fun in the five days spent preparing for it than during the actual show. □□□

5102 Vineland Avenue, North Hollywood, CA 91601. (213) 761-0511.



Note: Choir were miked with four U87s, while three C451s and two PZMs on Vega wireless units provided audience miking.

stationary studio might be booked for them. Weather can be both a problem, or a welcome diversion. If it's cold or rainy, the walk from the recording site to the mobile can be a drag. But, if it's a beautiful day, it can be an inspiration. Usually an enclosed walkway overcomes the problem days.

Towards the Future

Like everyone else in the studio business, most mobile owners await the "seasoning out" of digital audio. Many consider that it's too early to invest large amounts of money in one or more digital multitracks. [Although John Moran's Digital Services' mobile, equipped with dual Sony PCM-3324 digital multitracks, currently is out on the road — *Ed.*]

The need for a large number of inputs will always be great, and when digital consoles with higher price-tags come of age, mobile operators will be forced with a problem of economics. Most likely, the next generation of mobiles will have analog consoles routing to a digital storage medium, a situation that probably will remain for quite some time.

Improvements will come in terms of better audio monitoring, and special designs in console programming to easier facilitate the demands of mobile sessions. Who knows, maybe a wireless mobile unit will be developed! Me? I'm waiting for a mobile Space Shuttle. Ah, dream on. ■■■

ENACTRON TRUCK — Recording Quincy Jones' "Seattle Celebration"

At the Paramount Theatre, Seattle, Washington, on March 12, 1983, a band, an orchestra, a choir, an independent sound production company, a lighting production company, a video crew from a local television station, and a mobile recording truck all came together along with Patti Austin, James Ingram, and Ray Charles to celebrate Quincy Jones' 50th birthday. They called the event the "Seattle Celebration," and to record the concert Jones' staff enlisted the aid of the Enactron Truck.

The Enactron Truck is a 40- by 8-foot mobile recording studio, from within whose walls came Emmylou Harris' first eight albums, Willie Nelson's *Stardust*, and more. Among its other credits are the movie soundtracks for *The Last Waltz*, *The Rose*, and *A Star is Born*. Live performances recorded by the truck includes Linda Ronstadt, Styx, Black Sabbath, Joe Walsh, and James Taylor. Recording hardware centers around a Neve 36/24 console,

ROCSHIRE RECORDING: Mobile 1 is equipped with 3M M79 24-track and 2-track, and a Sony PCM-1610 digital processor; custom 32-input API console; monitors are modified JBL 4320s; full outboard rack of signal processors. **1240 North Van Buren, Anaheim, CA 92807. (714) 632-5046.**

CHATON RECORDINGS: two Otari MTR-90 16/24-track and MX5050B 2-track tape machines; Soundcraft Series 800 27 input/8 bus console +direct out from each channel; Tannoy SRM12B monitors; dbx 900 modular processing system; video equipped; **5625 E. Nauni Valley Drive, Scottsdale, AZ 85253. (602) 991-2802.**

Each mobile also must be designed around its own operational criteria. If the unit is designed to accommodate remote "studio" recording, it must have on-board cue system for use during overdubbing. Also important is the provision of a stereo speaker foldback system, so that an artist can hear a performance through a set of monitors without having to go back into the control room.

Mobile recording units come in all shapes and sizes. The largest ones, such as Westwood One and Enactron, are housed in large 45-foot tractor trailers, the advantage being that a lot of space is available to accent a comfortable working area. In the case of Westwood One, the "space age" interior features a bar and lounge that take up about a third of the unit. Artist interviews can be conducted in the lounge, and visitors shouldn't get in the way of work being done. Some mobiles are housed in large buses, as in the case of Mountain Mobile and Cleveland's Recording Connection. Sometimes sleeping quarters and a kitchen are added to larger units.

One problem with operating big mobiles can be the cost of transportation, which is reflected in a mileage charge of at least $1.00 per mile to the client. Le Mobile and The Record Plant use smaller trucks that provide better gas mileage, and fewer client luxuries, yet keep the working space to the maximum. Artisan and Full Sail Mobile are housed in motor coaches. From the outside, compared to other vehicles, these mobiles seem small, but the working area inside can be very spacious. An advantage is the ability to park anywhere, even at the artist's home, and never have the problem of becoming an eyesore with anxious neighbors. By comparison, fuel cost is less than half of the large-sized mobiles.

Mobile versus Stationary Studios

Obviously, if you're going to throw a concert and want it recorded, only a mobile can handle it correctly and economically. But what about recording a "studio" album with a mobile? First, the biggest advantage of using a mobile is versatility; the sonic choices are almost infinite. Want a large hall rhythm track? Then simply move the recording venue to a large hall. Want to get that feel of an artist playing in his own living room? Then record in his own living room. In fact, the choices are limited only by imagination, and access to mains power. By way of an example, Adrian Belew's recent LP, *Twang Bar King*, was recorded in different rooms to achieve various sound characteristics.

Another advantage is the facility's exclusivity. With mobile recording, no one unaffiliated with the project alters a microphone or console setup. And it's totally removed from any office staff and related personnel.

As far as a comparison of price is concerned, each situation dictates its own economics. Most mobiles quote a daily rate of between $2,000 and $4,000, and often require a half-day set up. But, like any studio operation, very few mobiles will be unaverse to giving handsome price breaks, depending on the amount of time involved and availability. Sometimes, if a well-known artist shows an interest the rate will drop, just to land recognizable references for future business. All in all, mobile studio recording can be a "best buy," if handled correctly.

Not to seem totally smitten by mobile studio recording, it should be acknowledged that a few problems can exist. Visual communication, such as non-verbal, producer-to-artist responses to takes, don't exist as they do in stationary-studio situations; the control room vibe can't be visualized "through the glass." Sometimes though, this may be an asset. Very often everything but lead vocals work well in a mobile, so a

DIGITAL SERVICES' MOBILE — *continued* . . .

ease of set-ups or mixdown. An auxiliary Neve console is available for expanding to all 54 active inputs, if necessary.

A large Audio + Design Scamp effects rack houses a variety of signal processors, including limiters, compressors, expanders, noise gates, delay lines, etc. Practically any brand of equipment requested is available with advance notice from the client. Reverb systems consist of a Sony DRE-2000 digital reverb, and a MICMIX two-channel spring reverb. An ADR Vocal Stressor also is included in the processing rack, and more equipment is being added on a regular basis.

Digital Services owns and operates the first two Sony PCM-3324 24-track digital machines in the U.S., and has had extensive experience with the digital Compact Disc format Sony PCM-1610 two-track processor and U-Matic combination, and its companion DAE-1100 digital editor. The company has provided digital recording, editing, and mixdown facilities to a wide variety of artists, including Frank Zappa, Neil Young, Dionne Warwick, the Chicago Opera, Joe Cocker, and Marshall Tucker. All digital equipment is available for use in the truck, as well as conventional Otari MTR-90 analog 24-tracks.

Built into the front wall over the console are a set of JBL 4311 monitors; auxiliary monitoring includes a Klein and Hummel tri-amp system, Ed Long Time Align™ MDM-4, Yamaha NS-10M, and Auratones. Between the 4311 audio monitors are located two video monitors, and the truck is equipped with its own video camera, or can accept a feed from a separate video truck. A total of 16 line-level outputs are available for sending a mono or stereo feed to other support systems, such as video/film trucks, broadcast booth, director, etc. In addition, there are specific, dedicated lines to handle SMPTE/sync code signals, without trying up one of the usual audio channels. The truck currently boasts an RTS intercom system, although a wireless system is currently under assessment. □□□

1001 River Oaks Banks Tower, 2001 Kirby Dr, Houston, TX 77019. (713) 520-0201.

— continued from page 81 . . .

can have no second take, maintaining a mobile is a big job. A good mobile operation will keep a check list that covers all equipment, and the vehicle itself, and rundown each item before and after each date.

Dealing with union stage, light, and audio crews can be critical when handling live dates. A good crew has the ability to deal with the situation on any level. For example, union crews can be hard-core about non-union people laying lines or carrying equipment, depending on the nature of the union in each particular city. (Some unions don't care who runs the lines or carries the gear, but in other towns the union may care a *lot*.) Like snake charmers — if you'll pardon the pun — a mobile's crew must be able to take a potentially lethal situation and make it tame, whatever it takes.

CRITERIA RECORDING: MCI/Sony JH-600 48 mike Input, 36 line Input/32-out console; two MCI/Sony JH-24 24-tracks and two JH-110B 2/4-tracks with Dolby and dbx noise reduction, Mitsubishi X-80 2-track and X-800 32-track Digital tape machines; monitors by UREI, JBL. Criteria Recording and Fanta Professional Services of Nashville, TN have recently entered into a joint venture agreement to market the Criteria mobile recording truck. All inquires for the mobile rig will be handled through Johnny Rosen of Fanta. **1213 16th Avenue South, Nashville, TN 37212. (615) 327-1731.**

Recording live sound is a totally different bag of tricks compared to a studio situation. Pressure runs high, and every aspect of the recording process is in constant motion, changing every minute. An engineer needs a cool head, and a tremendous sense of humor to get through it all. If something goes wrong, or is missed, there is *no* second chance. Solutions to quick and sometimes complex problems must be decided in a split moment.

An engineer recording a live concert must make allowances for level headroom once a concert actually starts. Sound checks are only good for about 70% of the actual level once the performers get on-stage in front of an audience. It's prudent to run your VU peaks to around -2 dB in rehearsal, because as soon as the band starts playing it's going to increase — I always keep in mind multitrack levels *first*, then the two-track reference mix. Preparation must be made for the incredible dynamics of front-line vocals; without at least some compression, serious distortion is at risk.

In the event that a sound check did not occur before a concert, the engineer usually sets all faders to zero or null, and quickly pads each mike trim across the console to optimum level as soon as the concert begins. Believe it or not, to the bewilderment of all involved in live recording, this happens all the time.

Although pressure runs high, so do the thrills; live-mix engineers experience more adrenalin in an hour than most people do in a week.

The Mobile Working Environment

Stepping into any of the new breed of mobile environments can be a heady time warp into the future, each resembling a mini "Battlestar Gallactica." Mobile owners spend a good buck to make sure the environment is functional, and very comfortable.

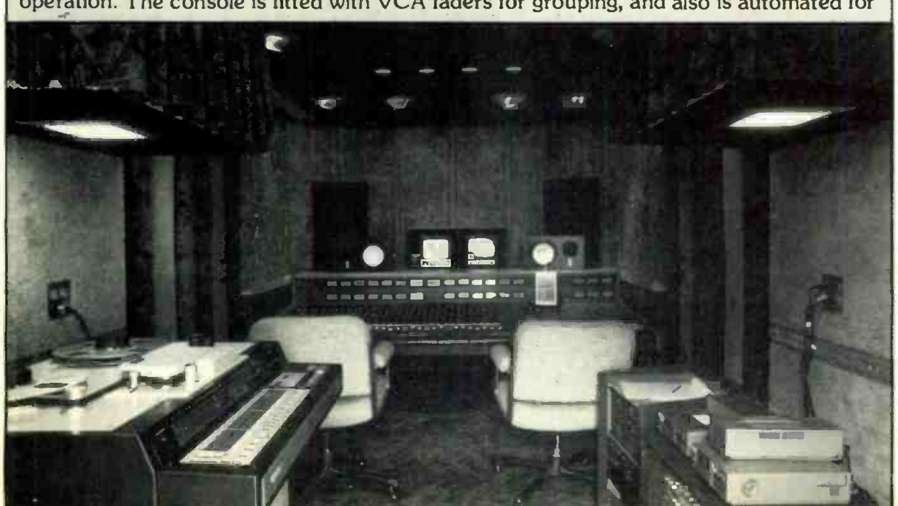
Special design considerations need to be taken into account, since the unusual stress patterns encountered on the road put a lot of strain on barrier walls and joints, as well as recording equipment. As a mobile tracks down a highway, it twists and flexes in a way that could tear apart interior walls if they are too rigid. Mobile designer Gary Hedden says that he computes these stress factors, and includes them in any construction process. "Building a mobile is much more like designing an aircraft, than a regular building," he offers. Air conditioning and heating must be clean and quiet, a feat not easily accomplished in some installations.

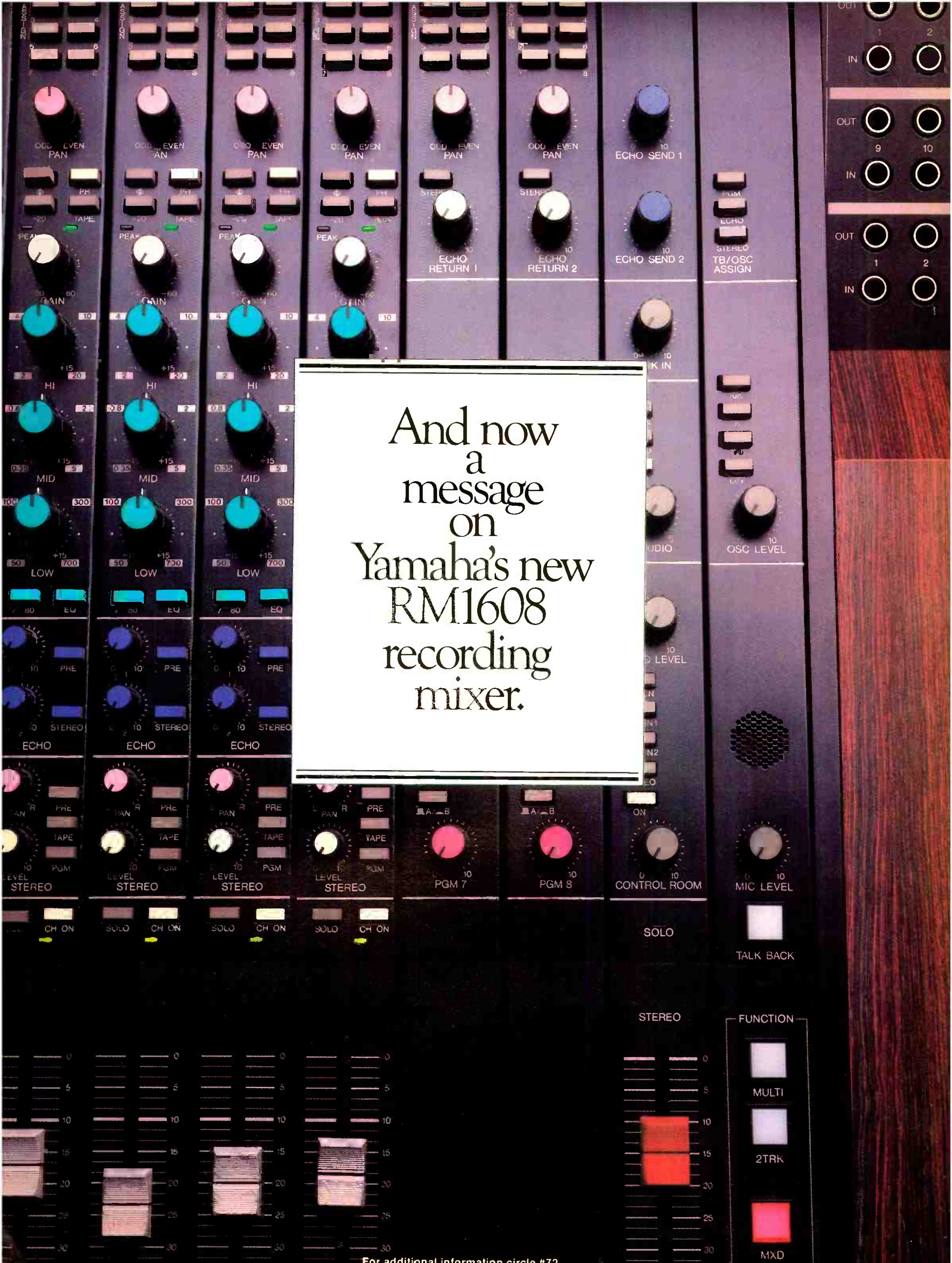
DIGITAL SERVICES' REMOTE VEHICLE — Equipped With Sony PCM-3324 Digital Multitrack

Digital Services' Audio Control Truck 1, based in Houston, Texas, comprised a Chevrolet C-50 chassis with an insulated 23-foot box built out as an audio control room. Air conditioning is supplied by a central heat/air Carrier unit of two-tons capacity, with a central thermostat in the control room. Interior lighting is provided by four separate dimmable circuits that provide light independently to all areas of the room; a 12-volt lighting system also is available to provide work light in all critical areas should external power be removed. These systems, as well as the refrigerator, are on their own transformer-isolated AC system, separate from the technical power system.

The main signal input comes from a 54-way stage box that utilizes Jensen transformers to produce three outputs for each microphone input: one direct out, and two transformer-isolated outputs. Both of the transformer outputs have separate ground lifts. Input to the truck from the stage box is via AMP multipin connectors.

The console is an MCI JH-636 transformerless design, currently set up for 32 input/dual 24 bus output. It is fitted with the MCI Broadcast Option, which allows separate microphone control to the tracking busses and the mix bus, as well as normal tracking/mixdown operation. The console is fitted with VCA faders for grouping, and also is automated for




And now a message on Yamaha's new RM1608 recording mixer.
For additional information circle #72

RM1608

SPECIFICATIONS

TOTAL HARMONIC DISTORTION (T.H.D.)
Less than 0.1% at +4dB *output, 20Hz to 20kHz (all Faders and controls at nominal)

HUM & NOISE (20Hz to 20kHz) Rs = 150 ohms
(INPUT GAIN "– 60")

– 128dB	Equivalent Input Noise (E.I.N.)
– 95dB	residual output noise: all Faders down.
– 80dB	(84dB S/N) PGM Master volume control at maximum and all CH PGM assign switches off.
– 64dB	(68dB S/N) PGM Master volume control at maximum and one CH Fader at nominal level.
– 73dB	(77dB S/N) STEREO Master Fader at maximum and all CH STEREO level controls at minimum level.
– 64dB	(68dB S/N) STEREO Master Fader at maximum and one CH STEREO level control at nominal level.
– 80dB	(70dB S/N) ECHO SEND volume at maximum and all CH ECHO volumes at minimum level.
– 75dB	(65dB S/N) ECHO SEND volume at maximum and one CH ECHO volume at nominal level.

CROSSTALK
– 70db at 1kHz: adjacent Input.
– 70db at 1kHz: Input to Output.

MAXIMUM VOLTAGE GAIN (INPUT GAIN "– 60")

PGM	74dB: MIC IN to PGM OUT.
	24dB: TAPE IN to PGM OUT.
	34dB: ECHO RETURN to PGM OUT.
	14dB: PGM SUB IN to PGM OUT.
STEREO	74dB: MIC IN to STEREO OUT.
	24dB: TAPE IN to STEREO OUT.
	34dB: ECHO RETURN to STEREO OUT.
ECHO	70dB: MIC IN to ECHO SEND.
C/R	74dB: MIC IN to C/R OUT.
	24dB: 2 TRK IN to C/R OUT.
STUDIO	74dB: MIC IN to STUDIO OUT.
	24dB: 2 TRK IN to STUDIO OUT.

CHANNEL EQUALIZATION
± 15 dB maximum
HIGH: from 2k to 20kHz PEAKING. MID: from 0.35k to 5kHz PEAKING. LOW: from 50 to 700 Hz PEAKING.

HIGH PASS FILTER – 12dB/octave cut off below 80Hz.

OSCILLATOR Switchable sine wave 100Hz, 1kHz, 10Hz

PHANTOM POWER 48V DC is applied to XLR type connector's 2 pin and 3 pin for powering condenser microphone.

DIMENSION (W x H x D) 37-1/2" x 11" x 30-1/4" (953 mm x 279.6 mm x 769 mm)

Hum and Noise are measured with a – 6dB/octave filter at 12.47kHz; equivalent to a 20 kHz filter with infinite dB/octave attenuation.
*OdB is referenced to 0.775V RMS.
• Sensitivity is the lowest level that will produce an output of – 10dB (245mV), or the nominal output level when the unit is set to maximum gain.
• All specifications subject to change without notice.

The specs speak for themselves. But they can't tell you how natural, logical and easy the RM1608 is to work. All the controls and switches are logically arranged to help you get the job done quickly and accurately.

And in the tradition of Yamaha's sound reinforcement mixers, the RM1608 sets new standards of reliability as well as ease of operation. For complete information, write: Yamaha International Corporation, P.O. Box 6600, Buena Park, CA 90622. In Canada, Yamaha Canada Music Ltd., 135 Milner Ave., Scarborough, Ont. M1S 3R1.

YAMAHA

For additional information circle #73

MISSISSIPPI DELTA BLUES Festival

form with frequency response throughout the 60 degree angle of coverage.

Power Amplifiers

The main system was powered by six CS-800 power amplifiers, which are rated at 400 watts into four ohms with both channels driven simultaneously. Three CS-800s (six channels) were used to drive the FH-2s on the low-end, one and a half CS-800s (three channels) to drive the mids (MB-1), and three channels to power the highs (CH-4C). The total available power from the six power amps was a conservative 5,000 watts for the whole main system.

The CS-800 amplifier features a patented DDT (Distortion Detection Technique) circuit, a type of dynamic compression that maximizes the performance of the amplifier/speaker combination by sensing conditions that might overload the amplifier, and activates compression only when clipping is imminent. The DDT compression circuit is triggered by the onset of voltage clipping, current limiting, or amplifier slew limiting.

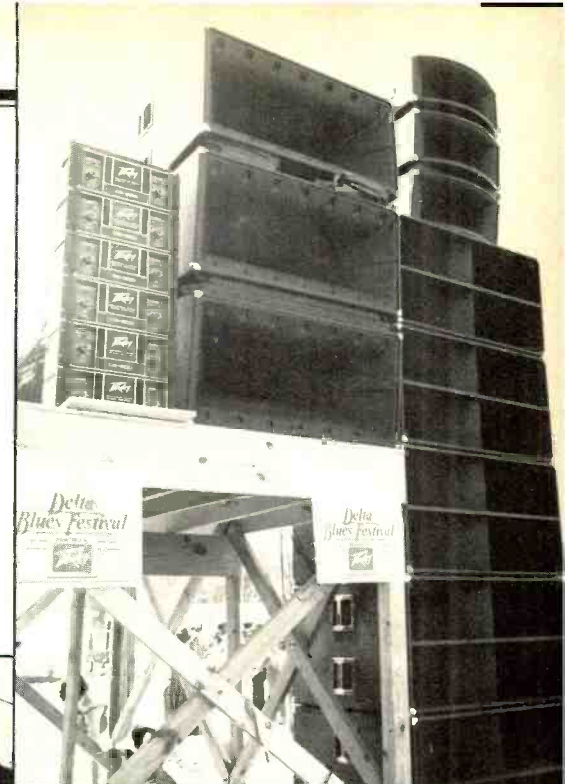
An obvious advantage of using power amplifiers equipped with such a device is that in a three-way system, for example, crossed over at 250 Hz and 1.2 kHz, each band (low, mid, and high) has its own *independent* compression circuitry that is activated only by the onset of clipping or distortion within each individual frequency band. By utilizing the conventional approach to a three-way system — with an outboard compressor inserted into the signal path before the crossover — the dynamic range of the *entire* system is affected as soon as the compression threshold is reached, no matter what type of signal reached the threshold of the compressor (kick drum, guitar, etc). Even by employing three outboard compressors dedicated for lows, mids, and highs inserted after the crossover in a system set up with a standard amplifier, Peavey claims that the unique performance of a three-way system powered by CS-800s with DDT compression *cannot* be duplicated.

Any normal application using an outboard (before the power amp) compressor will affect the dynamic range because the outboard compressor can only affect the line signal or input voltage to the power amplifier. Therefore, the compressor cannot know where the power amp's rails are, or what its full power capabilites are. With DDT in the amplifier, however, an input/output comparitor measures the input and output voltages, and only compresses the amp to its full power capability. (In other words, it compresses to wherever the rail voltage is.)

DDT also is said to eliminate listener fatiguing distortion because it *only* attenuates the amplifier to its full power output capability. Any compressor placed before the power amp, as opposed to being an integral component of the amp's circuitry, will offer more adverse limitations (if you'll pardon the pun) to the system's dynamic range.

Crossover Units

The crossover comprised an Electronic Crossover Mainframe with a PL-250 and a PL-1200EQ plug-in crossover module, making the crossover points 250 Hz and 1.2 kHz. In series with the low-out of the ECM (250 Hz and below) was a PL-Subsonic high-pass filter. The PL-250 and the PL-1200EQ each offer 18 dB per octave high- and low-pass rolloff,



Main speaker stack and associated power amps (top); first stack located to left of stage near near-field fills, plus amp rack for first stack and monitor system.

while the PL-Subsonic rolls off the system below 40 Hz at a rate of 24 dB per octave. The Subsonic filter is effective in eliminating stage rumble and vibrations, plus wind noises, and can even prevent system damage should a microphone stand accidentally be knocked over. The unit also minimizes the danger of equalizing the system below cut-off, which can happen frequently enough at an outdoor venue by a guest band's inexperienced soundperson, for example.

System Equalization

A Peavey EQ-27 third-octave graphic equalizer, White Model 140 third-octave real time analyzer, and micplexer with three AKG C451E microphones was used to equalize the system. After equalization (the broadest peak had been between 150 Hz and 250 Hz) system response was found to be within +2 and -3 dB from 45 Hz to 12 kHz. Further equalization was introduced to obtain a 3 dB per octave rolloff beginning at 2 kHz. This very gradual HF rolloff is effective in minimizing the effect of excessive highs that are accentuated by the use of dynamic microphones with their associated presence peak (rise in high-frequency response) in close-miking situations. For example, there are more highs four inches away from the bell of a trumpet than there are 10 feet away from it, and close miking causes even more highs to be amplified than the instrument's natural timbre; a 3 dB per octave HF rolloff results in a more natural and less strident high frequency response. (Of course, any channel in the mix needing additional presence can be equalized above this 3 dB per octave rolloff.)

Delay Speaker Stacks

Due to the venue's geography, with the stage set up in one corner of a rectangular field, two delay towers were necessary to provide adequate sound

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coverage for the rear of the audience area. The first tower was located to the left of the stage on a six-foot high scaffold, in order to cover the area to the left of the stage that was out of the pattern of the main system. (The main stack was able to cover about 80% of the audience.) The second delay tower was located 160 feet from the stage, and about 40 feet behind the mixer platform on a 12-foot high scaffold.

Both delay stacks consisted of various Project One Series speaker system components — first stack: FH-2, MB-1, and CH-4C; second stack: FH-2, MB-1, and MF1-X 90-degree constant directivity horn — powered by two CS-800 amplifiers. Each delay stack was equalized to have a flat power response, and a 3 dB per octave rolloff above 2 kHz.

View from the house mixing position to the stage, showing both main and small speaker stacks.

To determine the correct delay times, a Peavey Digital Delay Processor was used for each stack, and the delay set up using an Acoustilog Impulser. The latter unit emits a positive pulse of varying frequency, thereby allowing the operator to drive the main system and adjust the digital delay line while listening to both the main and delayed system from a location beyond the delay tower. (In most situations this will also require a tool that has been used by men for centuries — the good old hand signal.) The Digital Delay Processor is capable of providing a maximum delay of 1.13 seconds, with a full 20 Hz to 20 kHz signal bandwidth.

House Mixer

The mixing console used at the Delta Blues Festival was a 24-input Peavey Mark IV, which is said to have been designed specifically with live sound reinforcement mixing in mind. The Mark IV offers transformer-balanced inputs, pre- and post-fader effects sends and returns, two monitor sends, four bands of equalization, one effects send, and assignment switches, including a direct to Sum and a PFL. Each of the mixer's four submasters has a variable 12 dB per octave low-cut filter, and both switched and unswitched outputs. (The switched outputs enable a sub to be patched through an equalizer, effects device or other signal processor, and then returned to the Sum bus via the Sum Aux jack). Sub output #1 was used for vocals, Sub #2 guitars, Sub #3 organ, harmonica, sax, etc., and Sub #4 drums.

The Mark IV also is equipped with phantom powering, and a complete intercom and talkback facility.

An EQ-27 third-octave graphic equalizer was employed on the drum mix, and Peavey stereo graphic 10-band equalizers on vocals and instruments.

Monitor System

The monitor rig used two Project Two systems (biamped with two CS-800s) as cross-stage units. The Project Two system includes an FH-1 low-frequency enclosure, an MB-2 mid-bass horn, and an MF1-X HF horn; in addition, a total of 10 Model 1545 enclosures were available for floor monitors. The 1545 monitor features a 15-inch Black Widow/-Super Structure speaker, and a non-radial geometry HF horn on a Model 22-A compression driver. Six CS-800 power amplifiers were available for the

Up Your Drums!

Take control of your drum sound with ATM microphones by Audio-Technica. We have all the right models to deliver the sound that moves your group and your audience. Our ATM11R, for instance, goes down to 30 Hz and is crisp and smooth all the way up to 20 kHz. What you play is *exactly* what they hear! Move in as close as you wish without distortion or overload. The ATM11R can mike *anything* you play except the kick drum (and we've got a killer mike for that)!

The great sound and high headroom (141 dB SPL) of the ATM11R comes from our advanced Fixed-Charge electret element, remote-powered from your mixer. It has an ultra-thin diaphragm (just *4 microns* thick) to respond instantly and accurately to every sound you make.

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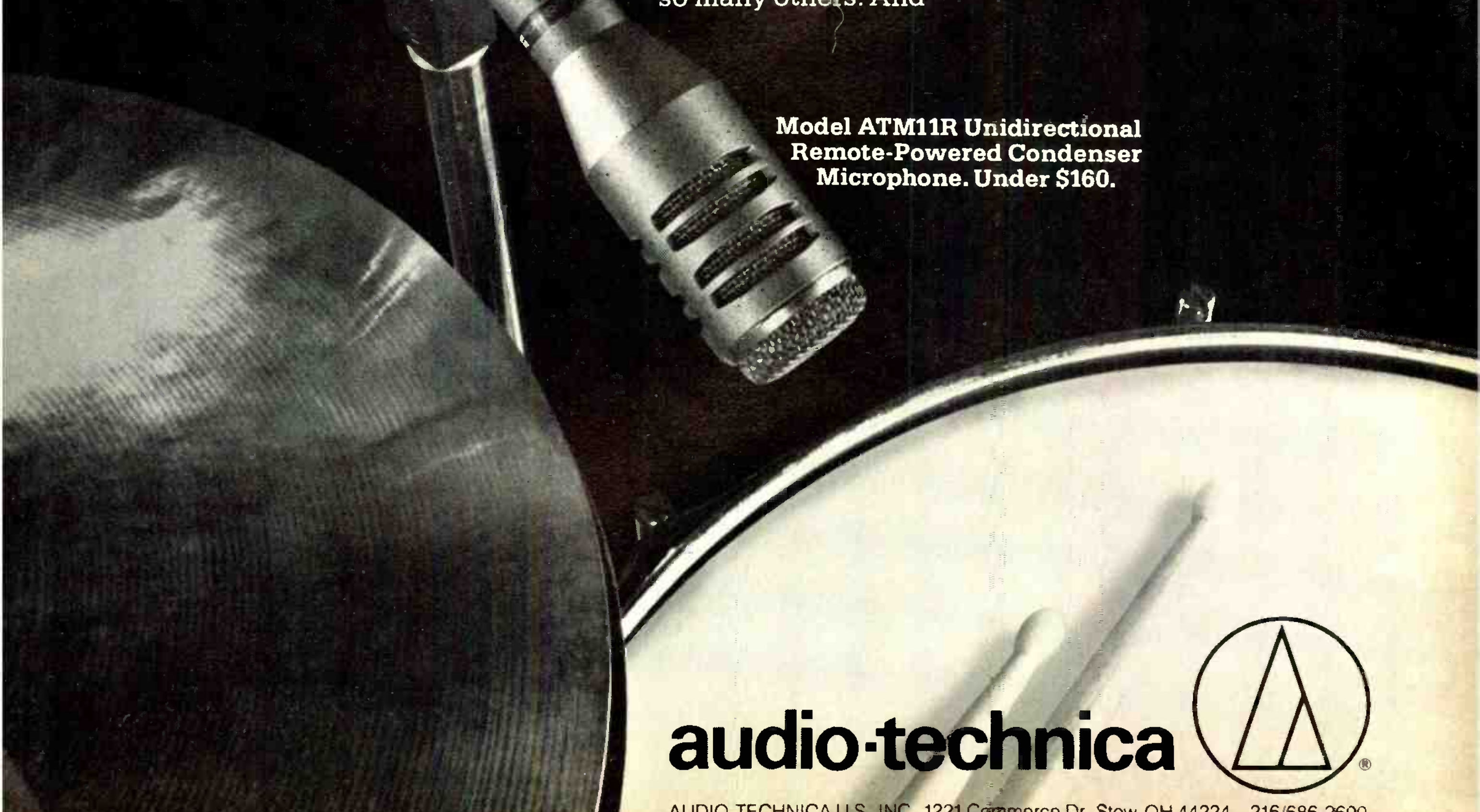

Model ATM11R Unidirectional Remote-Powered Condenser Microphone. Under $160.

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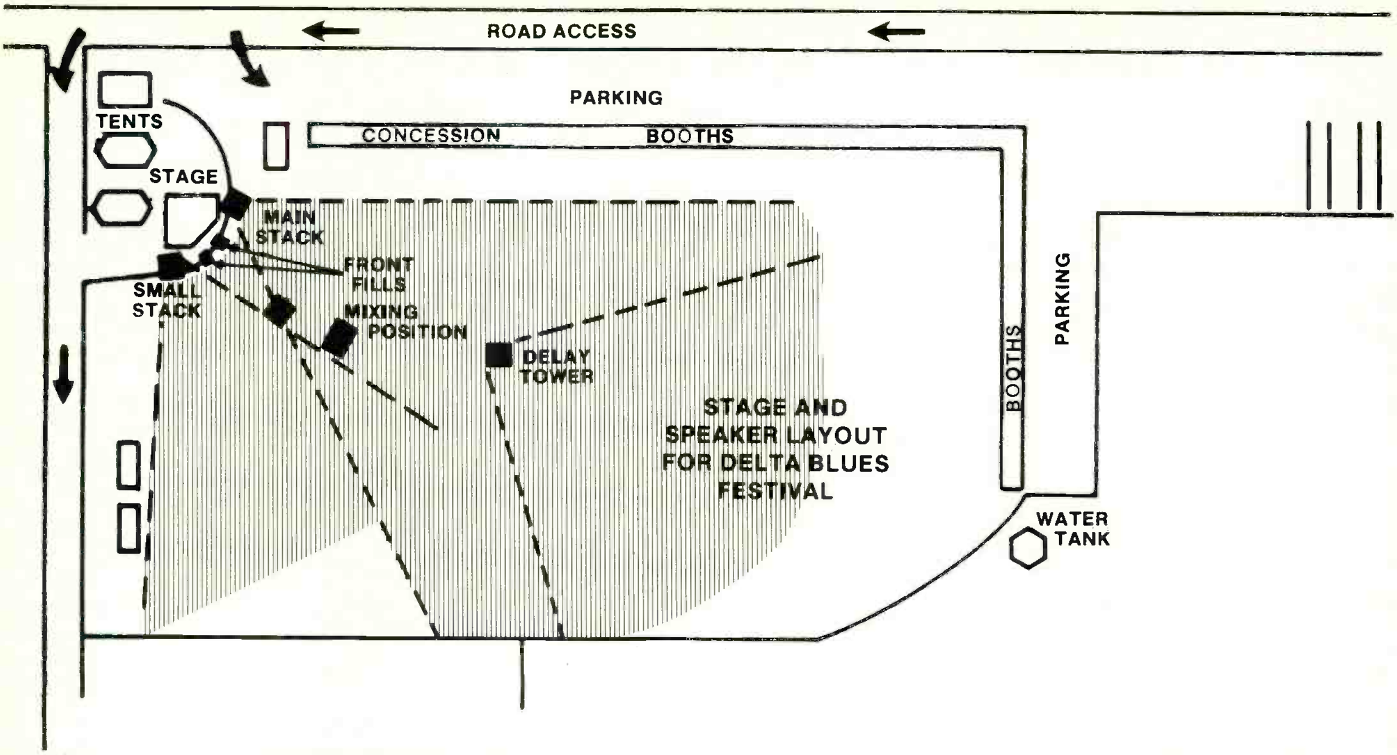


monitor mix, and eight EQ-27 third-octave graphic equalizers for monitor equalization.

The Mark IV 24/8 stage monitor mixer is identical to a Mark IV main console, with the addition of microphone splitters at the inputs, and an eight-way mixing matrix. The monitor mixer also boasts a unique channel patch feature (which is like a built-in direct box) enabling the engineer to patch into the monitor console unbalanced, and then send a balanced signal to the house mixer through the board's input transformer. The Mark IV has equalizer (or other processing) send and return patching for each monitor mix, which enables any of the eight mixes to be sampled after (post) that mix's equalization, through an identical monitor cabinet. Built-in intercom/talkback

MISSISSIPPI DELTA BLUES Festival

facilities provide total communication between the monitor mixer and the house position, or to any of the eight monitor mixes, and any number of additional remote "Talk Com" portable units.

Monitor mixes consisted of two separate cross-stage monitor feeds; a mix for the video crew and radio remote; a soloist mix; and up to four individual mixes for the floor monitors.

Monitor mixing position, with outboard signal-processor and EQ rack, and cross-stage monitor cabinets.

On-Stage Miking

All vocal microphones were Peavey Celebrity Series CD-30 dynamic cardioid models that incorporate an acoustic filter for controlling feedback. Drums were miked with both CD-30s and HD-40 hypercardioid microphones, which also were used to mike brass and reed instruments, high and low drivers of a Leslie cabinet; CD-30s were also employed on acoustic instruments and harp.

Electric guitars were taken direct from the speaker outputs of the backline amplifiers, using a Peavey EDI (Equalized Direct Interface) direct box. Conventional sound reinforcement practice allows two basic ways to route guitar, electric bass, and keyboard signal to the main PA system: close-microphone placement; or, if the amplifier/preamplifier is equipped with such a facility, direct output via a rear-panel connection. Both techniques, unfortunately, have severe drawbacks. Close mike placement creates an emphasizing effect at high frequencies which, in turn, generates an unnatural timbre. The direct out of the amplifier provides an accurate response of the instrument and amplifier, but bypasses a vital link in the audio chain — the loudspeaker.

Peavey's EDI direct box is said to provide all the advantages of the direct output but, through the use of different circuitry components, "synthesizes" the loudspeaker's response curve so that the timbre and tonality found in the main PA system is indistinguishable from the instrument/amplifier combination. The EDI box accomplishes this by controlling the resonant peak and high-frequency rolloff of the signal, in order to duplicate the voicing of the loudspeaker.

Miscellaneous Equipment

Because there was a crowd control area fenced in 30 feet around the front of the stage (in which sat the Governor and officials), two distributed and delayed SP-2 front-fill speaker enclosures were used to cover this area with a mix from the monitor console containing vocals, guitars, and any solos. The main array was positioned to begin covering the audience with mids and highs at a distance of 30 to 35 feet from the stage. A third SP-2 carrying the same mix was used to cover the area in which the artists lounged immediately backstage.

A couple of hours into the Blues Festival a fourth SP-2 cabinet was added to this backstage artist's mix in order to cover a number of fans that had gathered far to the right of the stage, out of the pattern of the mids and highs of

the main array. They were located so far to the right, in fact, that the stage wasn't even visible, but the crowd could see the artists as they arrived and departed. All four SP-2 cabinets were bi-amped using two CS-800 amplifiers, and a PL-800EQ plug-in crossover module. This "soloists" mix was rolled off below 150 Hz at 12 dB per octave using the variable low cut on the master module of the seventh monitor mix. The mix provided the necessary direct field of the mids and highs for these important areas not intended to be covered by the main system. Since the main system consisted of six FH-2s for the low-end below 250 Hz, there was no problem with low-frequency material being heard in these areas (the FH-2 generates 116 dB SPL at 1-watt/1-meter).

Peavey also provided a back line of instrument amplifiers for many of the musicians, including a Max Bass amp with two 3620 enclosures, three Heritages, two Austins, and an XR-600B with two bi-amped Model 118 International enclosures for keyboards.

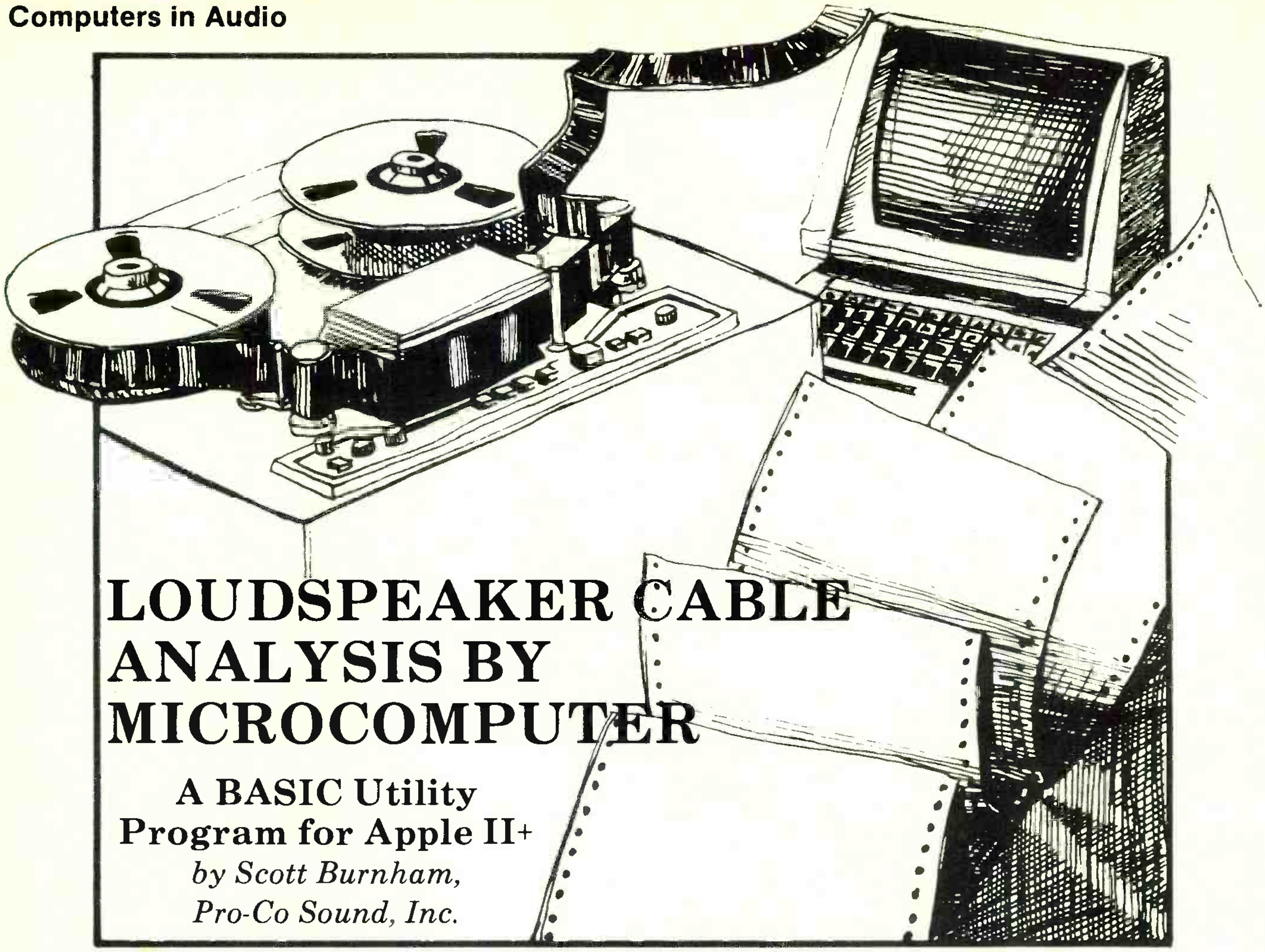
Performance — Festival Sound

The Delta Blues Festival got underway at noon on Saturday, September 17. The first couple of acts did not really show off the system's capabilities, due to the fact that they were a three-piece drum and fife band, followed by a solo performance on a "one-string" guitar (the original blues instrument). From then on the sound system really "wowed" the crowd, for the following acts featured contemporary instrumentation. The bass was considered by many to be deep, tight, and well defined, the mids smooth while retaining depth and instrument definition or imaging, and the high-end bright, articulate, and not harsh in anyway. After walking all around the cotton field, and listening from many vantage points, this writer considers that the Peavey single-source three-way system definitely provided the intended coverage.

The various artists on the Festival agenda offered a wide selection of Blues material, ranging from some of the softest ballads, with SPL measuring below 80 dBA at the mixer position (120 feet from the speakers), upwards through crescendos above 110 dBA during the more contemporary R&B numbers. The most demanding band that played through the system that afternoon was certainly The Nighthawks, who had traveled all the way from California to be a part of the Blues Festival. The soundman for The Nighthawks operated the mixer during the group's most demanding set, and remarked that the system was capable of providing "excellent sound!"

The "Mojo Magic" of the Mississippi Delta Blues Festival had some 30,000 blues fans clapping and dancing, and also consuming barbecue, chitterlings, fish sandwiches, hot tamales, and Polish sausages. The consensus of opinion was that this year's sound by Peavey was by far the best of the Festival's six-year history. Malcolm Walls, public relations director for the Mississippi Action for Community Education remarked that "the quality of the sound brought the whole festival alive. The sound just flowed and was heard equally well everywhere."

The Delta Blues Festival presented an opportunity for Peavey Electronics to show that their Project One Series speaker system could prove to be an example of economy of scale. Dozens of sound companies would have attacked this exercise in sound reinforcement with multiple stacks of equipment. Peavey's approach of employing one single-source array for 80% of the required coverage demonstrated that excellent quality sound is possible from a system of modest size, weight, and price. ■■■



LOUDSPEAKER CABLE ANALYSIS BY MICROCOMPUTER

A BASIC Utility Program for Apple II+

by Scott Burnham, Pro-Co Sound, Inc.

Ask any sound technician, installer, or engineer to describe the equipment in a particular audio system, and the chances are that one critical component will not even be mentioned: the cable. While cables can dramatically affect the performance of a system, all too often they are selected without any regard for the loss, distortion, and noise they may contribute to a signal.

Proper loudspeaker cables are a particularly critical component in a well-designed audio system. Speaker cables affect system performance in two major ways: power loss, and damping factor. The former probably sounds more important, but the latter actually has more effect in terms of perceived audio quality. These two effects are closely related, in that both result from the impedance of the cable connecting the amplifier to the speaker.

The impedance of the cable in series with that of the speaker forms a power divider; Ohm's Law predicts that the higher the cable impedance, the lower the power that reaches that speaker itself, and the greater the amount of power wasted as heat dissipated by the cable. Furthermore, the damping factor also is dependent on the ratio of the source impedance (the amplifier and cable) to the load impedance (the speaker). Damping factor is the measure of the amplifier's ability to "damp" or suppress unwanted movement of the speaker cone. Good damping is vital to accurate sound reproduction; poor damping causes "muddiness," and lack of definition in bass instruments.

Calculating power loss and damping factor requires a wire resistance table, and the proper application of Ohm's Law plus a few other formula. Although this approach yields accurate results, it is, at best, tedious. Another method is to use a nomograph, a sort of "slide rule" that relates source and load impedance, cable gauge, resistance and length, and resulting damping factor. Such nomographs often are found in the owners' manuals of professional-type power amplifiers, as an aid to cable selection. Nomographs get the job done, but the results are fairly approximate.

A better solution? Use a computer, of course!

SCAM (Speaker Cable Analysis for Microcomputers) is a computer program developed to calculate various parameters of system performance with regard to the speaker cables being used. As can be seen from the accompanying listing written in Applesoft BASIC, the program provides the user with two different options for analysis and design: Option #1 computes power loss and damping factor, given amplifier damping, load impedance, and cable gauge and length; while Option #2 allows the user to specify "target" value of damping or power loss, then choose either the cable gauge to be used, or the length necessary. Having selected the required values, SCAM then "fills in the blanks." (For instance, the user may specify a maximum power loss of 10%, a load impedance of 4 ohms and a cable gauge of #16; SCAM will then provide the resulting damping factor of 8.26, and maximum cable length of 55.28 feet.)

SCAM has been found to be a great labor-saving tool and conversation piece for audio techno-freaks and engineers. Although written in Applesoft BASIC for the Apple II+ microcomputer, it can be easily adapted to other computers using different versions of BASIC simply by modifying a few statements peculiar to the Apple. For instance, the screen clearing routine for the Apple uses a HOME or CALL-936 statement; the printer/display select routine uses PR#1 to turn on the printer connected to peripheral slot number 1; PR#0 to turn it off; and the GET statement, which is similar to MBASIC's "INPUT$," might need to be changed to suit the reader's system.

When entering SCAM from the computer keyboard, be very careful to type all lines exactly as listed, including spaces and punctuation. The REMarks are not necessary to program execution, but make modification much easier by clari-

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LOUDSPEAKER CABLE ANALYSIS BY COMPUTER

fying program logic and organization. The program as listed is completely functional and almost crash-proof. If you have any problems, carefully recheck for spelling, syntax, or punctuation errors.

* * *

For more information concerning speaker cables and their effects upon amplifier and speaker performance, the following references may prove helpful:

Electronic Wire and Cable, Catalog 882; Belden Corporation.

Handbook of Electronic Tables and Formula; Howard W. Sams & Co., Inc., 1979.

Institute of High Fidelity, *Official Guide to High Fidelity*; Howard W. Sams & Co., Inc., 1974.

Instruction Manual IM-3A, Crown International.

Ray Kimber, "Speaker Wire"; *R-e/p*; April 1980.

Harry Maynard, "Speaker Cables: How Various Types Affect Sound"; *Radio-Electronics*; December 1978.

Owner's Manual, Model 750B & C Professional Power Amplifier; BGW Systems.

A.N. Thiele, "Loudspeakers in Vented Boxes, Part II"; *Journal of the Audio Engineering Society*: Volume 19, No. 6; June, 1971.

— SAMPLE PRINTOUT —

```
*** COMPUTED WITH SPEAKER CABLE
ANALYSIS PROGRAM BY
PRO-CO SOUND, INC.
AMPLIFIER DAMPING AT 8 OHMS: 200
SPEAKER LOAD IMPEDANCE: 16 OHMS
AMPLIFIER DAMPING AT 16 OHMS: 400
CABLE LENGTH: 20 FEET
CABLE GAUGE: 12
RESISTANCE: 1.59E-03 OHMS/FOOT
TOTAL RESISTANCE OF CABLE: .0636 OHMS
POWER LOSS IN CABLE: .4 %
DAMPING FACTOR: 154.44
```

APPLESOFT LISTING OF SCAN (SPEAKER CABLE ANALYSIS FOR MICROCOMPUTERS) PROGRAM CONFIGURED TO RUN ON APPLE II+ PERSONAL COMPUTER

```
10 REM SPEAKER CABLE ANALYSIS
11 REM FOR MICROCOMPUTERS
12 REM
13 REM A/K/A "SCAM"
14 REM APPLE II PLUS VERSION
15 REM
20 REM BY S. R. BURNHAM
21 REM DIRECTOR OF ENGINEERING
22 REM PRO-CO SOUND, INC.
23 REM
25 REM COPYRIGHT (C) 1983
26 REM PRO-CO SOUND, INC.
27 REM KALAMAZOO, MI. 49007
28 REM
30 REM MAY 1983
31 REM REVISED JUNE 1983
32 REM
33 REM
40 TEXT : HOME : CLEAR
50 D$ = CHR$ (4): REM CTRL-D
60 I = 8: REM NUMBER OF GAUGES "ON FILE"
70 DIM G(I): REM GAUGE ARRAY
90 DIM R(I): REM OHMS-PER-FOOT ARRAY
95 REM EACH GAUGE IS FOLLOWED BY ITS OHMS-PER-FOOT RESISTANCE
100 DATA 8,.000628
110 DATA 10,.000999
120 DATA 12,.00159
130 DATA 14,.00252
140 DATA 16,.00402
150 DATA 18,.00639
160 DATA 20,.0101
170 DATA 22,.0162
180 DATA 24,.0257
185 REM READ DATA INTO ARRAYS
190 FOR X = 1 TO I
200 READ G
210 G(X) = G
240 READ R
250 R(X) = R
260 NEXT X
270 RESTORE
275 REM SET UP SCREEN
280 PRINT "******** SPEAKER CABLE ANALYSIS ********"
290 PRINT
300 PRINT "THERE ARE TWO OPTIONS AVAILABLE. "
310 PRINT
311 PRINT
320 INVERSE : PRINT "OPTION #1";: NORMAL
321 PRINT " COMPUTES AMPLIFIER DAMPING"
322 PRINT "AND SPEAKER CABLE POWER LOSS, GIVEN THE LOAD IMPEDANCE
 AND THE LENGTH AND GAUGE OF THE CABLE."
330 PRINT
340 INVERSE : PRINT "OPTION #2";: NORMAL
341 PRINT " COMPUTES MAXIMUM CABLE"
342 PRINT "LENGTH OR SPECIFIES A MINIMUM GAUGE,    GIVEN EITHER T
HE MAXIMUM POWER LOSS     OR THE MINIMUM DAMPING FACTOR ALLOWED."
350 PRINT
351 PRINT
360 PRINT "PRESS <1> OR <2>, OR <RETURN> TO EXIT: ";
380 GET A$
390 IF A$ = "1" THEN GOSUB 430: GOTO 40
400 IF A$ = "2" THEN GOSUB 690: GOTO 40
410 IF A$ = CHR$ (13) THEN GOTO 1880: REM ASCII RETURN
420 GOTO 380
430 REM OPTION #1
440 HOME
450 PRINT "OPTION #1"
460 GOSUB 1440: REM GET AMPLIFIER DAMPING AND LOAD IMPEDANCE
470 HOME
480 PRINT "ENTER LENGTH OF CABLE IN FEET"
481 INPUT "AND PRESS <RETURN>: ";L$
490 L = VAL (L$)
500 IF L < = 0 THEN GOTO 470
510 HOME
520 PRINT "DATA AVAILABLE FOR EVEN GAUGES ";G(1);"-";G(I)"."
530 PRINT
535 PRINT
540 PRINT "ENTER THE GAUGE OF THE CABLE"
541 INPUT "AND PRESS <RETURN>: ";AWG$
550 AWG = VAL (AWG$)
555 REM SEE IF GAUGE IS "ON FILE"
560 FOR X = 1 TO I
570 G = G(X)
580 IF AWG = G THEN GOTO 620: REM GAUGE IS "ON FILE"
590 NEXT X
600 GOTO 510
620 R = R(X): REM GET OHMS-PER-FOOT
630 RW = (L * 2) * R: REM WIRING RESISTANCE
640 PL = (RW / (RW + RL)) * 100: REM POWER LOSS
650 RO = 8 / DF: REM AMPLIFIER OUTPUT IMPEDANCE
660 D = RL / (RO + RW): REM DAMPING FACTOR
670 GOSUB 1570: REM DISPLAY RESULTS
680 RETURN
690 REM OPTION #2
700 HOME
710 PRINT "OPTION #2"
```

. . . continued overleaf —

APPLESOFT SCAN PROGRAM LISTING — *continued . . .*

```
720  GOSUB 1440: REM  GET AMPLIFIER DAMPING FACTOR AND LOAD IMPEDA
NCE
730  HOME
740  PRINT "PRESS <P> TO SPECIFY"
741  PRINT "MAXIMUM ALLOWABLE POWER LOSS"
750  PRINT
760  PRINT "PRESS <D> TO SPECIFY"
761  PRINT "MINIMUM ALLOWABLE DAMPING FACTOR"
770  PRINT : GET A$
780  IF A$ = "P" THEN  GOSUB 910: GOTO 810
790  IF A$ = "D" THEN  GOSUB 980: GOTO 810
800  GOTO 730
810  HOME
820  PRINT "PRESS <L> TO SPECIFY"
821  PRINT "LENGTH OF CABLE"
830  PRINT
840  PRINT "PRESS <G> TO SPECIFY"
841  PRINT "GAUGE OF CABLE"
850  PRINT : GET A$
860  IF A$ = "L" THEN  GOSUB 1050: GOTO 890
870  IF A$ = "G" THEN  GOSUB 1250: GOTO 890
880  GOTO 810
890  GOSUB 1570: REM  DISPLAY RESULTS
900  RETURN
905  REM  MAXIMUM POWER LOSS ROUTINE
910  HOME
920  PRINT "ENTER THE MAXIMUM POWER LOSS"
921  INPUT "ALLOWABLE (%) AND PRESS <RETURN>: ";PL$
930 PL =  VAL (PL$)
940  IF PL < 0 OR PL >  = 100 THEN  GOTO 910
950 RW = (PL * RL) / (100 - PL): REM  WIRING RESISTANCE
960 D = RL / (RO + RW): REM  DAMPING FACTOR
970  RETURN
975  REM  MINIMUM DAMPING ROUTINE
980  HOME
981  PRINT "DAMPING FACTOR SPECIFIED MUST BE GREATERTHAN ZERO AND
LESS THAN AMPLIFIER      DAMPING AT SPEAKER LOAD IMPEDANCE."
982  PRINT : PRINT "AMPLIFIER DAMPING                   AT SPE
AKER LOAD IMPEDANCE: ";DL
983  PRINT
984  PRINT
990  PRINT "ENTER THE MINIMUM DAMPING FACTOR"
991  INPUT "ALLOWABLE AND PRESS <RETURN>: ";MD$
1000 D =  VAL (MD$)
1010  IF D >  = DL OR D <  = 0 THEN  GOTO 980
1020 RW = (RL / D) - RO: REM  WIRING RESISTANCE
1030 PL = (RW / (RW + RL)) * 100: REM  POWER LOSS
1040  RETURN
1050  HOME
1055  REM  PICK THE GAUGE ROUTINE
1060  PRINT "ENTER LENGTH OF CABLE IN FEET"
1061  INPUT "AND PRESS <RETURN>: ";L$
1070 L =  VAL (L$)
1080  IF L <  = 0 THEN  GOTO 1050
1090 R = (RW / L) / 2: REM  FIGURE TARGET OHMS-PER-FOOT
1095 R =  INT (R * 1000000 + .5) / 1000000: REM ROUND TO TRICK APP
LESOFT BUG
1100  FOR X = I TO 1 STEP  - 1: REM  START AT SMALLEST GAUGE
1110  IF R(X) <  = R THEN  GOTO 1180: REM  SEE IF ITS OHMS-PER-FOO
T IS LESS THAN TARGET
1120  NEXT X
1125  REM  ERROR MESSAGE
1130  PRINT : PRINT "LENGTH SPECIFIED REQUIRES A CABLE GAUGE LARGE
R THAN ";G(1);
1140  PRINT " TO ACHIEVE POWER LOSS"
1150  PRINT "OR DAMPING FACTOR REQUIRED."
1160  PRINT : PRINT "PRESS <RETURN> TO EXIT
"
1165  PRINT : PRINT "PRESS ANY OTHER KEY TO
CONTINUE ";: GET A$
1166  REM  ESCAPE CLAUSE USES "POP"
1167  IF A$ =  CHR$ (13) THEN  POP : RETURN

1170  GOTO 1050
1180 G = G(X)
1190 R = R(X)
1200 RW = R * L * 2: REM  WIRING RESISTANCE
1210 D = RL / (RO + RW): REM  DAMPING FACTOR

1220 PL = (RW / (RW + RL)) * 100: REM  POWER
 LOSS
1240  RETURN
1250  HOME
1260  PRINT "DATA AVAILABLE FOR EVEN GAUGES
";G(1);"-";G(I);","
1265  REM  MAXIMUM LENGTH ROUTINE
1270  PRINT
1275  PRINT
1280  PRINT "ENTER GAUGE OF CABLE"
1281  INPUT "AND PRESS <RETURN>: ";AWG$
1290 AWG =  VAL (AWG$)
1295  REM  SEE IF GAUGE IS "ON FILE"
1300  FOR X = 1 TO I
1310 G = G(X)
1320  IF AWG = G THEN  GOTO 1350
1330  NEXT X
1340  GOTO 1250
1350 R = R(X): REM  OHMS-PER-FOOT
1360 L = (RW / R) / 2: REM  LENGTH
1390  RETURN
1435  REM  AMPLIFIER DAMPING AND LOAD IMPEDA
NCE ROUTINE
1440  PRINT
```

```
1441 PRINT : PRINT "DAMPING FACTOR IS TYPICALLY FOUND IN THETECHN
ICAL SPECIFICATIONS FOR THE POWER  AMPLIFIER."
1442 PRINT : PRINT "IT IS USUALLY SPECIFIED AT 8 OHMS. IF  SPECI
FIED AT 4 OHMS, MULTIPLY BY 2. IF  SPECIFIED AT 16 OHMS, DIVIDE BY
 2."
1443 PRINT : PRINT "A TYPICAL VALUE IS 200 AT 8 OHMS. "
1444 PRINT
1445 PRINT
1450 PRINT "ENTER SPECIFIED DAMPING FACTOR"
1451 PRINT "OF AMPLIFIER AT 8 OHMS"
1452 PRINT "AND PRESS <RETURN>"
1453 PRINT "(IF UNKNOWN, PRESS <RETURN>"
1454 INPUT "TO ENTER DEFAULT VALUE OF 200): ";DF$
1460 IF DF$ = "" THEN DF = 200: GOTO 1490
1470 DF =  VAL (DF$)
1480 IF DF <  = 0 THEN  HOME : GOTO 1450
1490 RO = 8 / DF: REM  AMPLIFIER OUTPUT IMPEDANCE
1495 RO =  INT (RO * 1000 + .5) / 1000: REM  ROUND TO TRICK APPLES
OFT BUG
1500 HOME
1510 PRINT "ENTER SPEAKER LOAD IMPEDANCE IN OHMS"
1511 INPUT "AND PRESS <RETURN>: ";RL$
1520 RL =  VAL (RL$)
1530 IF RL <  = 0 THEN  GOTO 1500
1540 DL = RL / RO: REM  AMPLIFIER DAMPING AT LOAD IMPEDANCE
1545 DL =  INT (DL * 1000 + .5) / 1000: REM  ROUND TO TRICK APPLES
OFT BUG
1560 RETURN
1570 HOME
1571 L =  INT (L * 100 + .5) / 100
1572 PL =  INT (PL * 100 + .5) / 100
1573 D =  INT (D * 100 + .5) / 100
1574 REM  PRECEDING FORMATS OUTPUT OF SOME PARAMETERS
1579 REM  PRINTOUT ROUTINE FOR SILENTYPE PRINTER
1580 PRINT "DO YOU WANT A PRINTOUT OF THE RESULTS?"
1590 PRINT
1600 PRINT "PRESS <Y> OR <N> ";: GET A$
1610 PRINT
1620 IF A$ = "Y" THEN  PRINT D$;"PR#1": GOTO 1642: REM  TURN ON P
RINTER
1630 IF A$ = "N" THEN  GOTO 1650
1640 GOTO 1600
1642 PRINT : PRINT "*** COMPUTED WITH SPEAKER CABLE ANALYSIS PROG
RAM BY PRO-CO SOUND, INC."
1643 PRINT
1644 PRINT
1650 HOME
1660 PRINT "AMPLIFIER DAMPING AT 8 OHMS: ";DF
1670 PRINT
1680 PRINT "SPEAKER LOAD IMPEDANCE: ";RL;" OHMS"
1690 PRINT
1700 PRINT "AMPLIFIER DAMPING AT ";RL;" OHMS: ";DL
1710 PRINT
1720 PRINT "CABLE LENGTH: ";L;" FEET"
1730 PRINT
1740 PRINT "CABLE GAUGE: ";G
1750 PRINT
1760 PRINT "RESISTANCE: ";R;" OHMS/FOOT"
1770 PRINT
1780 PRINT "TOTAL RESISTANCE OF CABLE: ";RW;" OHMS"
1790 PRINT
1800 PRINT "POWER LOSS IN CABLE: ";PL;" %"
1810 PRINT
1820 PRINT "DAMPING FACTOR: ";D
1830 PRINT
1840 PRINT
1850 PRINT D$;"PR#0"
1860 PRINT "PRESS ANY KEY TO CONTINUE";: GET A$
1870 RETURN
1880 END
```

□□□

New Products

URSA MAJOR ANNOUNCES STARGATE 323 STEREO DIGITAL REVERB

The StarGate 323 is described as an extremely flexible device with eight room-simulations, including tiny chambers, fast-diffusing plates, concert halls, and huge echoing spaces. Rooms can be modified by front-panel controls that adjust decay time, pre-delay, and separate high- and low-frequency decay curves.

The unit is designed to be user-friendly, so that the operator whose experience extends only to conventional plate and spring reverbs will understand its operation immediately. Digital readouts on the front panel show decay time, pre-delay, and room selection, while eight discrete LEDs monitor signal level. There is also an input level control, an input-mute, reverb-clear, and dry-only buttons, each of which can be operated by foot pedals.

At the same time, the company says, engineers familiar with first-generation digital reverbs will be able to use the 323's advanced features to their full advantage. The unit features a 15 kHz bandwidth, which is evident even at the longest decay times, and a dynamic range of 80 dB.

Pre-delay offers 16 choices, from zero to 320 milliseconds, while decay time has 8 choices, ranging from zero to 10 seconds, depending on the selected room mode.

Professional net price of the Stargate 323 is $2,500.

URSA MAJOR, INC.
BOX 18
BELMONT, MA 02178
(617) 489-0303

For additional information circle #84

ELECTRO-VOICE EXPANDS DL SERIES WOOFER LINE

"Designed with different bass responses, enclosure-size requirements, and target applications, the new DL12X and DL18X, like their predecessor the DL15X,

perform at the high-efficiency end of the direct-radiator spectrum [on the order of 5%]," explains Jim Long, director of marketing professional sound reinforcement products. "The DL series is suited for a broad variety of vented and horn enclosures."

According to Long, the woofers' carefully engineered drive system assures high efficiency, linear, low-distortion output, and high power capacity. The low-mass voice coils are made of rectangular aluminum wire, edge-wound on a rugged laminated polymide form. A break-up resistant diaphragm and suspension ensure a smooth, musical upper-bass sound and plenty of low-frequency shock capability or "punch."

The DL drive system is augmented by two exclusive E-V features: the Thermal Inductive Ring (TIR™) and its PROTEF™ coating. TIR provides a significant heat-transfer path from the top of the voice coil, while simultaneously moderating the normal inductive rise of the voice coil. PROTEF, a Teflon-based protective coating, is applied to the inside diameter of the top plate, adjacent to the voice coil, and provides protection by lubricating any rubbing contact, and inserting electrical insulation between the coil and the top plate.

The DL12X driver is suited to midbass applications in three- or four-way systems, or as a woofer in two-way systems where response to 70 Hz is appropriate and a small enclosure is required. Specifications include a frequency response (in a 2.6-cubic-foot vented enclosure tuned to 52 Hz) of 58 Hz to 5.2 kHz, ±3 dB; a 350-watt long-term average power capacity per AES recommended practice (100 Hz to 1 kHz); and a sensitivity at 1 meter, 1 watt (average from 200 Hz to 4 kHz) of 100 dB.

Designed for sub-woofer use in three- or four way systems, the D18X offers the highest output in the lowest octaves because of its large cone area. Specifications include a frequency response (in the 13-cubic-foot TL405 vented enclosure) of 36 Hz to 3 kHz, ±4 dB; a 500-watt long-term average power capacity per AES recommended practice (100 Hz to 1 kHz); and a sensitivity at 1 meter, 1 watt (average from 200 Hz to 4 kHz) of 99 dB.

The DL Series woofers have pro-user net prices of $192.00 (DL12X), and $320.00 (DL18X).

ELECTRO-VOICE, INC.
600 CECIL STREET
BUCHANAN, MI 49107
(616) 695-6831

For additional information circle #85

ELECTRO-VOICE BK-1 "BLACKNIGHT" CONDENSER MIKE

The new BK-1 microphone departs from EV tradition in a number of respects, including its unique physical profile and black design, while combining much-desired features from other mikes in the company's line. Also dubbed the "Black Knight," the new cardioid condenser mike is said to be especially suited to the needs of vocalists, but also excellent for other applications. A useable bass-boosting proximity effect tailors low-frequency response with changes in the working distance.

The BK-1's high sensitivity, smooth peak-free frequency response and cardioid pickup pattern are said to assure excellent rejection of unwanted off-axis and reflected sounds which cause feedback in a

live entertainment situation. Power for the electret condenser-type mike can be from either by battery or phantom power; an on-off switch is provided to greatly prolong battery life.

ELECTRO-VOICE, INC.
600 CECIL STREET
BUCHANAN, MI 49107
(616) 695-6831

For additional information circle #86

COMPACT MODEL FB31 ENG/EFP MIXER FROM SHURE

The new FP31 mixer, which measures approximately 2 by 6¼ by 6 inches deep, included three locking-type inputs located for easy access on the mixer's left side, corresponding to the input level control side of the mixer. A slide switch located below each input connector allows the user to choose either mike- or line-level signals. The unit also includes two three-pin output connectors on the unit's right side, both of which are switchable for either mike- or line-level operation. A master control knob on the front panel sets the combined output level.

Front-panel features include a VU meter and lamp switch; overload/limited (LED) indicator; low-cut filter switches; limiter in/out switch; power and battery-check switches; tone oscillator switch; plus stereo headphone mini-jack and level control.

Headphone outputs can be used as additional unbalanced line feeds for connection to external equipment.

Also incorporated in each FP31 is a built-in slate microphone for voice announcements and emergency field use. The microphone is controlled by a momentary-contact pushbutton switch that also activates a timed (two-second) low-frequency slate tone.

Other features include a Simplex (phantom) or A-B power switch for use with condenser microphones; a tape-out jack for connection to a cassette deck; a 12 VDC power connector; and a battery compartment that accommodates three standard 9-volt batteries.

According to Shure, the mixer has been designed to provide a wide, flat frequency response and low distortion up to +18 dBm output. In addition, the unit operates with extremely low internal noise or external interference problems, and the switchable low-cut filters for each input effectively reject low-frequency handling and wind noises.

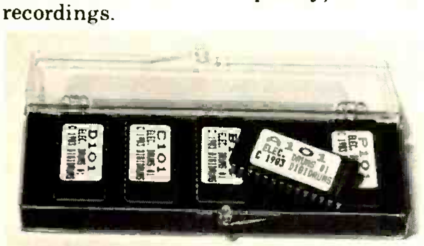
User net price of the FP31 mixer is $830.

SHURE BROTHERS INC.
222 HARTREY AVENUE
EVANSTON, IL 60204
(312) 866-2553

For additional information circle #87

DIGIDRUMS ANNOUNCES ALTERNATIVE SOUND CHIPS FOR DRUMULATOR

The alternate drum and percussion sounds for E-mu Systems' Drumulator consists of five EPROMs, containing an entire set of new, digitally recorded sounds to replace the standard Drumulator sounds. Sound sets currently available include electronic drums, Latin percussion, African percussion, heavy rock drums, jazz drums, sound effects and analog drum machine sounds. All Digidrums sounds are studio quality, low-noise recordings.



EPROM sets are easily interchanged — just remove the original Drumulator chips from their mounting sockets, and plug in a Digidrums set. No soldering or wiring is required. Each set of EPROMs includes a "program" chip containing software that determines the length of each sound, permitting longer, sustained sounds to be programmed. The program chip also contains updated E-mu software that increases the Drumulator's song storage capacity to 64 songs.

DIGIDRUMS
100 SOUTH ELLSWORTH
SAN MATEO, CA 94401
(415) 579-1514

For additional information circle #88

BRYSTON UNVEILS MODEL 2B-LP POWER AMP

The Model 2B-LP low profile, 50-watt per channel amplifier is designed to save space in recording studio and broadcast monitor applications, and is said to be especially applicable where difficult loads,

high average levels, and extreme musical accuracy are required.

Among the amp's standard features are balanced inputs with XLR-female connectors; dual-color pilot-clipping LEDs; extreme current delivery capability (24 amperes peak per channel); and front-removable channel-cards for near-zero downtime in field-servicing.

The 2B-LP can be considered, the manufacturer says, as the functional equivalent of the original Bryston 2B in half the space.

BRYSTONVERMONT LTD.
R.F.D. #4, BERLIN
MONTPELIER, VT 05602
(802) 223-6159

For additional information circle #89

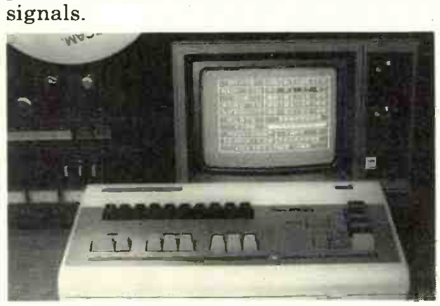
New Products

SYNCHRONOUS TECHNOLOGIES INTRODUCES LOW COST TIMECODE SYSTEM

Designed specifically for the smaller recording studio, the SMPL System provides a SMPTE timecode generator, timecode reader, Autolocator, automatic record in/out insert editing system, timecode-derived metronome, 24 tick/beat drum and synthesizer synchronizing system, and recorder remote control, as well as other features. At $995 price of the entire system is less than the cost of a typical timecode generator alone.

The SMPL System is designed to be used with lower cost multichannel cassette or open-reel machines, where it simply plugs into the normal remote-control jack. Neither tachometer output nor speed control input to the recorder are required.

With the system, insert editing no longer requires the combined skills of engineer, musician, and juggler. During rehearsal, punch-in and -out points can be set on the fly, and saved in the computer's memory to be repeated as many times as necessary. Separate Rehearse and Take modes allow the user to rehearse and preview the edit points as many times as necessary before committing to tape. A programmable Cue point provides both looping-type Return-to-Cue function at the end of the insert, and also provides a known, repeatable starting point for instrument synchronizing signals.



Many benefits result from the system being based on industry standard non-drop timecode. Unlike simple tone or click-track based instrument synchronizing systems, the SMPL System need not be started from the beginning of the work in progress, in order for the metronome and synchronizing signals to be in correct time-phase with the music.

With the new system, tapes produced in the small studio will transport to larger studios, and be compatible with automatic mixdown and sync-locking equipment. Tapes produced on machines with limited channels can be "pyramided" to larger-format multitrack machines, allowing the artist to work in his own environment at his own pace, and still have access to more expensive studio facilities.

Since much of today's commercial music involves digital drums and sequencer-controlled polyphonic synthesizers, the SMPTE track can replace numerous tracks which might otherwise be recorded as audio. Not only does this effectively increase the number of available tracks, but it also allows these tracks, which are susceptible to the quality loss from ping-ponging and conventional dubbing techniques, to be mixed first generation to the master tape.

SMPL also uses the "user bits" in the timecode to provide an indelible slate that becomes part of the control track, and which can be used as a cross reference for lead and track sheets, billing notes, etc.

SYNCHRONOUS TECHNOLOGIES
P.O. BOX 14467
OKLAHOMA CITY, OK 73113
(405) 842-0680

For additional information circle #92

BIAMP SYSTEMS INTRODUCES MODEL 44 KEYBOARD MIXER

The new Model 44 compact four-channel board features ultra-low noise, high slew rates, and low distortion without overload. Designed primarily for keyboard sound mixing and monitoring, the 44 also incorporates a high-quality microphone mixer for sound reinforcement.

Each of the four inputs channels has two high-gain instrument inputs, designed to prevent overload, plus a microphone input. A special Rhodes input on channel #4 is specially designed to balance the complete scale range without use of an external graphic equalizer.

All channels feature master effects send, independent effects patch in/out jacks, monitor send, three-band equalization, output level, and input overload LED. The returns section includes auxiliary input level to main, effects return to monitor, effects return to main, and tape inputs to main/monitor. The monitor system is wired post-EQ, pre-level, for use as both keyboard and microphone monitor mix.

BIAMP SYSTEMS, INC.
P.O. BOX 728
BEAVERTON, OR 97075
(503) 641-6767

For additional information circle #93

E-MU SYSTEMS UNVEILS SERIES OF OPTIONS FOR DRUMULATOR

The Drumulator Pad Programmer allows the digital drum machine to be played and programmed with drum sticks on four dynamically responsive pads, which should simplify the process of creating rhythms with natural sounding

accents. Sensitivity and accent level controls allow the response of each pad to be individually tailored to the user's playing style. In addition, the Programmer includes four trigger outputs for controlling synthesizers, sequencers or other drum machines, as well as four external controller inputs.

The Graphic Rhythm Controller is a software system for the Apple II or IIe that allows non-realtime programming of measures and songs with total dynamic control on a note by note basis. The GRC is composed of four linked program modules: with the Measure Writer module, a cursor is moved around a graphic representation of a measure of music, placing notes on the desired beats and setting their volume levels; Segment Writer and Song Writer modules are used to link measures into complete songs, and then The Song Player module is used to actually play the song on the Drumulator.

A selection of alternate sounds are also now available for the Drumulator. A user-installable EPROM containing a crash cymbal sound is available as a replacement for the standard ride cymbal.

E-MU SYSTEMS, INC.
2815 CHANTICLEER
SANTA CRUZ, CA 95062
(408) 476-4424

For additional information circle #94

SESCOM SB-1 MK II STEREO BUFFER AMP

Designed to provide a convenient means to interface a semi-professional, unbalanced VTR or tape recorder to balanced equipment, the gain of amplifier sections is adjustable from 0 dB to +30 dB via controls easily accessible from the front panel. The loss of the pad sections are fixed at -14 dB.

Other specifications include input impedance of 600 ohms balanced (pad section), and 100 kohms unbalanced (amplifier section); -101 dB noise below rated output; frequency response ±1 dB, 20 Hz to 20 kHz; distortion less than 0.2% at 20 Hz maximum rated output; and output 600 ohms balanced (amplifier section), and 600 ohms unbalanced (pad section).

Suggested retail price is $190.

SESCOM, INC.
1111 LAS VEGAS BLVD. NORTH
LAS VEGAS, NV 89101
(702) 384-0993

For additional information circle #95

ADA ANNOUNCES NEW MODEL 2.56I DIGITAL DELAY

The Model 2.56i produces delay times from 300 microseconds for high flanging, to over 2.5 seconds for digital tape loop effects — all at a full 16 kHz bandwidth. An LED rate indicator blinks at a rate equal to the delay time for accurate, real-time echo setting.

For producing special effects, the 2.56i has Regeneration, Modulation, and Repeat Hold features. The Regeneration Hi-Cut control reduces the high-frequency content in the delayed audio signal as it recirculated for a natural decay, and is variable between 16 kHz and 1.0 kHz. The Modulation section has an 8:1 sweep range that produces flanging that sweeps over three octaves.

The Waveform control continuously blends the shape of the modulation from a triangle to a sine to a square-wave. In addition, Modulation LEDs provide a visual indication of the speed, direction, and location of the sweep to its upper and lower limits. The Repeat Hold feature enables up to a 2.56-second musical segment to be sampled and repeated indefinately for background rhythm effects.

The Model 2.56i has a suggested list price of $799.95.

ADA SIGNAL PROCESSORS
2316 FOURTH STREET
BERKELEY, CA 94710
(415) 548-1311

For additional information circle #96

New Products

V-40 FULL-RANGE SYSTEM FROM CERWIN-VEGA

The complete V-40 consists of the B-215 (direct radiating, twin 15-inch bass) combined with a RMH-1000 (fully encased JMH-1 high-compression driver, with aluminum radial horn.) Since the system does not require bi-amplification to achieve maximum performance, two V-40 systems can be driven by a single dual-channel amplifier.

The system's direct radiating design is said to assure alignment of acoustic sources so that sound energy is propagated as a single wavefront, without smearing, over a time period. As a result, reproduction of percussive and other similar transient sounds are well defined, while music and vocals take on a more natural sound. The V-40 is described as being particularly well suited for playback applications involving music, or for motion picture theater sound systems.

The B-215 in the V-40's LF section employs twin, vertically-arranged 15-inch woofers in a direct radiating vented enclosure, tuned maximally flat to 35 Hz. The RMH-1000 HF compression driver and horn section is a fully encased version of the JMH-1 16-ohm, 50W one-inch throat compression driver mounted on an 800 Hz aluminum 90- by 45-degree radial horn. The cast aluminum horn is a straight exponential radial type with a nominal geometry of 90H by 40V degrees, and a theoretical cutoff frequency of 800 Hz. The horn provides optimum loading characteristics for the JMH-1 compression driver, with wider controlled dispersion over the entire bandwidth.

The V-40 crossover is a passive 1 kHz, 12 dB per octave highpass filter optimally matched to the system components. The network was designed for low loss, high power handling and low distortion at high input levels, and preserves dynamic range.

Suggested retail price for the V-40 is $940.

CERWIN-VEGA
12250 MONTAGUE STREET
ARLETA, CA 91331
(818) 896-0777

For additional information circle #99

SEQUENTIAL CIRCUITS INTROS SIX-TRAK SYNTHESIZER

The Six-Trak is described as a multi-timbral polyphonic synthesizer that lets the user play six completely different instrument sounds at one time, in a choice of three modes. One way is by layering several instrument sounds on top of each other, so that they can be played in unison from the keyboard.

Another way of playing the instrument as a multi-timbral "ensemble" is by recording six different instrument sounds into the Six-Trak's on-board multitrack digital recorder. Functions include record, playback, programmable playback speed, programmable track volume changes, variable-resolution error-correct, track duplication, and over 800-note storage capacity.

A third method for creating complex, multipart music is by combineing the live playing capabilities of the instrument with its multitrack recording feature.

The Model 610 Six-Trak is a six-voice polyphonic, fully programmable synthesizer featuring six VCOs and six, 4-pole filters. The synthesizer program memory stores 100 programs, each consisting of 33 voice parameters. Programs can be copied; the non-volatile memory is retained when power is off via a 10-year backup battery.

The back panel has jacks for audio output, a multipurpose control footswitch, and MIDI input and output. MIDI allows the integration of a Six-Trak into one programmable system, including SCI's new Drumtraks and Model 64 MIDI sequencer.

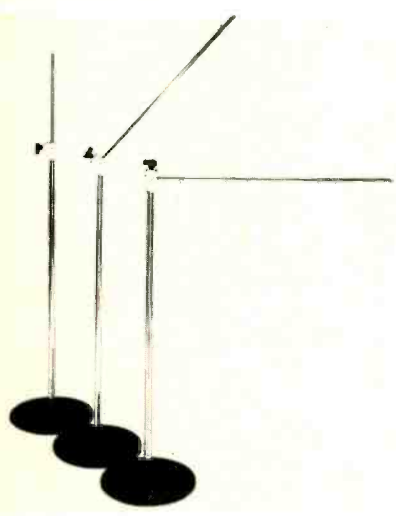
Suggested retail price of the Model 610 Six-Trak is $1,095.

SEQUENTIAL CIRCUITS, INC.
3051 NORTH FIRST STREET
SAN JOSE, CA 95134
(408) 946-5240

For additional information circle #100

NEW MICROPHONE SUPPORT FROM ATLAS SOUND

The MSB-1 incorporates the features of both a vertical microphone stand and a horizontal boom in one device. It is vertically adjustable in the standard height range — 42 to 70 inches — of a conventional mike stand. But, in addition, it easily converts into a 42-inch high floor stand with a 30-inch long horizontal boom by simply extending it to full height and repositioning the knob on the patented swivel. Because this is an integrated device, it requires no assembly or mechanical handling prior to use.



Tube assemblies are cold-rolled steel, heavily chrome plated. The upper tube is terminated in a ⅝-inch by 27 thread for all U.S.-standard microphone holders.

For extra stability, the MSB-21 is supplied with a one-piece, 12-inch diameter, extra-weight cast base equipped with anti-tip stabilizers and self-leveling shock absorbing pads.

ATLAS SOUND
10 POMEROY ROAD
PARSIPPANY, NY 07054
(201) 887-7800

For additional information circle #101

LOFT INTRODUCES MODEL 405-M MONO CROSSOVER

The Model 405-m five-way, 18 dB per octave crossover features front-panel,

recessed, crossover controls: Low 20 to 400 Hz; Low/Mid 100 Hz to 2 kHz; High/Mid 600 Hz to 12 kHz; and High 1.2 to 24 kHz. These crossover points allow the unit to be used as a three-way or four-way mono crossover. Also on the front panel are individual output controls, a power switch with LED indicator, and peak signal indicators for each frequency band.

Rear-panel signal connections are ¼-inch phone jacks (standard), or XLR connectors (optional). Suggested retail price is $599.

PHOENIX AUDIO LABORATORY
91 ELM STREET
MANCHESTER, CT 06040
(203) 649-1199

For additional information circle #102

New Products

NEW BOUNDARY ACCESSORIES FOR CROWN PZM® MIKES

Because of the PZM's hemispherical polar pattern, in some applications the mike may pick up too much unwanted sound, such as audience noise, muddy room acoustics, or squealing feedback. To solve such problems, Crown has introduced three new "boundaries" or panels to make the PZM polar response directional.

The A240 Boundary is a transparent plexiglass panel, two feet square by ¼-inch thick, to which a stand-mounted PZM can be attached. The boundary extends the low-frequency response, increases gain-before-feedback, and reduces pickup of leakage, audience noise, and room acoustics. Besides enhancing LF response, for a fuller bass sound, the boundary also makes the microphone sensitive to sounds approaching from the front of the panel, while rejecting sounds from the rear. By helping the PZM to reject audience noises, leakage (off-mike sounds from other instruments), and room acoustics, the mike can be turned up louder in sound-reinforcement systems before feedback occurs. The A240 includes an adjustable stand adapter to mount the panel to any standard microphone stand. Holes in the panel let the user suspend or "fly" the microphone/panel on nylon lines. Two microphone mounting clips are attached to opposite sides of the panel for stereo recording.

The Model 1560 and 1590 Isoflectors are L-shaped plexiglass boundaries to which a 6LP or 6S PZM can be attached. Designed for use on lecterns, the boundary increases the PZM's gain-before feedback, and reduces pickup of audience noise and room acoustics. The Isoflector is placed or mounted on the edge of the lectern farthest from the talker. The clear boundary is

nearly invisible from a distance, and is said to help reduce "mike fright" because there is no microphone pointing at the user's face. The Model 1560 Isoflector has a 60-degree angle between boundaries, while the Model 1590 Isoflector has a 90-degree boundary angle. Since the 1560 is more directional than the 1590, it is more effective in reducing feedback. However, the 1590 allows the talker more freedom of movement without getting "off-mike."

CROWN INTERNATIONAL
1718 W. MISHAWAKA ROAD
ELKHART, IN 46517
(219) 294-5571

For additional information circle #105

For additional information circle #104

QUAD-EIGHT/WESTREX ANNOUNCES NEW WESTAR PRODUCTION CONSOLE

Called Westar 1, the new model was designed to provide the user with a "World Class" console at an affordable price and, in keeping with this concept, is a full featured 24-track, automatable music recording and TV post production board.

Westar 1 features five different mike preamps and three different equalizers that plug in directly from the top panel, eight auxiliary sends, fader reverse, two stereo mix busses, and up to eight audio groups. All this in a modular frame that easily expands in the field.

The company will be displaying its new product at the up-coming AES/Paris, NAB/Las Vegas, and at AES/Anaheim conventions.

QUAD EIGHT/WESTREX
11929 VOSE STREET
NORTH HOLLYWOOD, CA 91605
(213) 764-1516

For additional information circle #106

SOUND CODE SYSTEMS 2350 POWER AMPLIFIER

To ensure years of dependable, clean power, a completely discrete design with MOS-FET output section has been incorporated into the Model 2350. MOS-FETs were chosen for their positive temperature coefficient, which provides for excellent longevity, the company says.

To further enhance the unit's reliability, a multispeed forced-air cooled heatsink with over 644 square inches of cooling is standard. To reduce mechanical noise, the air is directed into two large filtered inlet ports on the front panel of the amplifier prior to entering the heatsink assembly. Clipping indicators sense clipping in either the positive or negative output stages, not just one or the other, which is common in many other amplifiers.

The power supply section with large computer-grade filter capacitors and high current rectifier offer ample current minus objectionable ripple. This power supply in combination with the MOS-FET output section will deliver 260 watts into an 8-ohm load (350 watts into 4 ohm) both channels driven with less than 0.1% THD. The signal to noise ratio and slew rate have been optimized due to the lack of integrated op-amps in the audio chain.

Net professional price of the Model 2350 is $595.

SOUND CODE SYSTEMS
P.O. BOX 2198
GARDEN GROVE, CA 92642
(714) 554-0903

For additional information circle #107

STEWART ELECTRONICS MODEL ADB-1 ACTIVE DI BOX

Besides having exceptional frequency response, low distortion and low noise characteristics, wide dynamic range versatility is assured by an input that will handle 6-volt peak-to-peak, in the instrument position. When in the speaker position, the ADB-1 will handle up to 400W RMS, and allow a more accurate reproduction than conventional miking by running a signal directly off the speaker input.

Operating flexibility is assured by use of either a self-contained 9-volt battery (on-off switch is provided for long battery life), or by running on any phantom supply from nine to 60 volts.

Phase distortion and hum pickup are eliminated by a transformerless design, and annoying noise or radio transmission is minimized by RF protection on both input and outputs.

The Model ADB-1 chassis is said to be indestructable, and all switches are slide-type for maximum reliability. All connection and switch locations are clearly identified on the chassis top.

Suggested retail price is $119.50.

STEWART ELECTRONICS
P.O. BOX 60317
SACRAMENTO, CA 95860
(916) 929-4431

For additional information circle #108

New Products

YAMAHA INTRODUCES EMX LINE OF POWERED MIXERS

The EMX150, EMX200 and EMX300 are equipped with six, eight and 12 channels respectively, with the EMX150 rated at 150 watts per channel, and both the EMX200 and EMX300 at 250 watts. All units feature active input circuits for matching with any type of input, two monitor busses, one echo/effects bus, and stereo power amps rated at four ohms. All have three-band equalization with sweepable midrange and built-in analog delay system, as well as faders and interstage patching, and a stereo, nine-band graphic equalizer on the output.

The mixers directly address a major problem area for many integrated mixer/amplifier units — insufficient gain. In the EMX Series, the power amplifier is isolated from the low-level pre-amplifier stages, eliminating feedback from crosstalk. As a result, more than 90 dB overall gain is available from mike input to power-amp output.

Suggested retail prices are $1,195 for the EMX150, $1,595 for the EMX200, and $1,995 for the EMX300.

YAMAHA COMBO PRODUCTS
P.O. BOX 6600
BUENA PARK, CA 90622
(714) 522-9134

For additional information circle #112

SANKEN CU-41 TWO-WAY CONDENSER MICROPHONE

Until now, the basic performance dilemma of conventional single condenser capsule microphones has been that, in order to cover the full audio frequency range, they suffer reduced sensitivity. The 48-volt phantom powered CU-41 consists of one small diameter condenser capsule and one large diameter condenser capsule vertically mounted beneath a protective grill. The small one picks up the upper range of audio frequencies, and the large one the lower range. This two-capsule configuration is said to enable the CU-41 to have a wider frequency response without sacrificing its overall sensitivity.

Diameters of the two capsules are designed to provide an optimum balance between overall sensitivity and self-noise level. A newly designed electrical circuit combines the two outputs from the condenser capsules to produce a frequency response as flat as ±1 dB from 20 Hz to 20 kHz.

Acoustic design of the two condenser capsules is said to result in very uniform polar patterns over the audio frequency range. In addition, the two condenser capsules are made of a metal membrane, which makes the performance of the CU-41 unaffected even in a high humidity environment, and more immune to temperature changes.

PAN COMMUNICATIONS, INC.
MIYATA BUILDING
5-72-6 ASAKUSA
TAITO-KU, TOKYO 111, JAPAN
03-871-1370

For additional information circle #113

LOGITEK PAI-4 PRO AUDIO INTERFACE

The new unit allows unbalanced audio equipment, including VTRs and consumer gear, to be connected properly to balanced professional systems, and includes all balancing, impedance and level changing circuitry for interfacing two audio record lines and two audio playback lines properly, without degrading audio quality. Front-panel playback level controls can be used to control the audio output level of a VTR, allowing video editing without the use of an external audio mixer.

The PAI-4 fits in a 1¾-inch rack height space. All balanced connections are made via rear-panel XLR connectors, while unbalanced feeds come in and out on phono (RCA) jacks. Clear labelling make the unit easy to hook up, even for non-engineers.

Suggested net user price is $360.

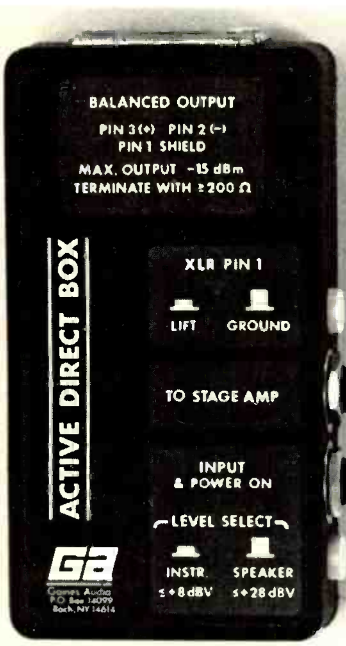
LOGITEK ELECTRONIC SYSTEMS
3320 BERING DRIVE
HOUSTON, TX 77057
(713) 782-4592

For additional information circle #114

GAINES AUDIO INTRODUCES AD-1 ACTIVE DIRECT BOX

The AD-1 may be used with any electric guitar, bass, synthesizer or other electronic instrument, and features a very high impedance input that will not load or otherwise affect the sound of any instrument. The output is an active balanced (transformerless) type at mike level, and will drive long cable runs without signal loss.

The circuit is said to offer low noise and distortion, flat frequency response, long battery life, and immunity to RF noise and magnetic fields. It is protected against input overloads and output short circuits, and is compatible with any audio system, including 48-volt, phantom-powered mike lines.



A ground lift switch allows the user to float pin #1 of the output XLR connector, eliminating ground loops. A Level-Select switch allows the unit to accept a wide range of inputs, from instrument level to line level to speaker level.

GAINES AUDIO
P.O. BOX 14099
FEDERAL STATION
ROCHESTER, NY 14614

For additional information circle #115

BIAMP EQ/290 THIRD-OCTAVE EQUALIZER

Designed for EQ adjustment to fine-tune room acoustics, ranging from live performance and studios, to fixed sound installation, front panel controls include subsonic 18 dB per octave filter, plus an adjustable high-frequency rolloff filter from 3.5 to 20 kHz.

Other features include floating and balanced circuitry on all inputs and outputs; 10-segment LED ladder; smooth, accurate combining action of filters; ¼-inch phone/XLR balanced/RCA plug for line input; overload peak indicator; and ground strap lift.

BIAMP SYSTEMS, INC.
P.O. BOX 728
BEAVERTON, OR 97075
(503) 641-6767

For additional information circle #116

REPLACEMENT TAPE HEADS FROM SPRAGUE MAGNETICS

The new line of ¼- and half-inch replacement tape heads for MCI JH-110A or -110B machines are manufactured by Woelke, West Germany. Sprague also has a complete line of erase heads for Studer, Otari, MCI, and many other decks.

Retail prices for ¼-inch erase, record or replay heads are $180, and half-inch are $380 each.

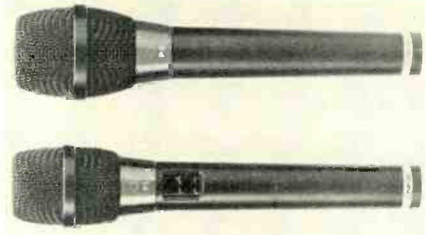
SPRAGUE MAGNETICS, INC.
15904 STRATHERN ST, #12
VAN NUYS, CA 91406
(818) 994-6602

For additional information circle #117

New Products

FENDER P-SERIES CONDENSER MICROPHONES

The two new mikes in the P-Series are said to provide the accurate response characteristics of high-performance recording microphones with ruggedness and reliability usually associated only with dynamic designs. According to division marketing director Steve Woolley, the new P-1 and P-2 cardioids incorporate significant innovations in their intended application area. "We don't give anything away to the best dynamic road microphones in areas like reliability and high SPL capability," he asserts. "For example, the P-1 can withstand 152 dB for 1% distortion, plus it has that precise, neutral sound and flat response of a high-quality studio condenser mike."



The P-2 includes a switchable low-frequency rolloff to compensate for the excessive bass "proximity effect" encountered in some close-miked vocal applications, while the P-1 incorporates a presence lift switch that adds a gently rising HF response for vocal applications, and a high pass/low cut switch for controlling bass response where desired.

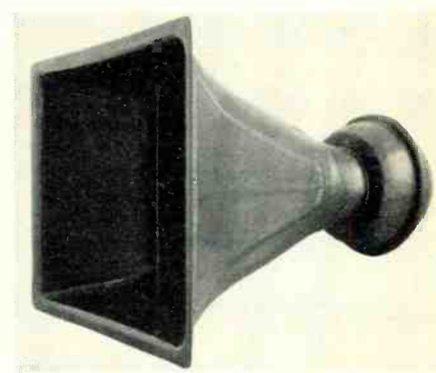
Both models may be powered by internal batteries, while the P-1 may be phantom powered for additional headroom. The P-1 carries a suggested retail of $220, while the P-2 is priced at $99, including case and swivel stand adaptor.

FENDER PRO SOUND
1300 E. VALENCIA DRIVE
FULLERTON, CA 92631
(714) 879-8080

For additional information circle #120

MODEL CB594 PATTERN CONTROL BASS HORN FROM COMMUNITY

With its mouth area of 16 square feet and a 50-inch air column length, the CB594 is described as being probably the only true, straight bass horn available today. The horn's high directivity and long throw capabilities is said to make it extremely useful in situations requiring controlled LF projection down to 50 Hz.



The CB594's pattern control features prevent midrange beaming, and affords a proper directivity match to midrange or high-frequency horns. Horizontal coverage is 90 degrees at 500 Hz, and 60 degrees at 800 Hz. The unit will accept either one 18-inch loudspeaker (for use in three-way systems, crossing over around 350 Hz into the M4 midrange) or one 15-inch driver (for use in two-way systems).

The CB594 is hand-laminated of weather-proof fiberglass; there are said to be no materials in the horn that can rust, corrode, or otherwise deteriorate due to weathering. The rear chamber containing the loudspeaker is provided with double-sealed neoprene gasketing, and is watertight. With compression chamber the horn is 68¼ inches long.

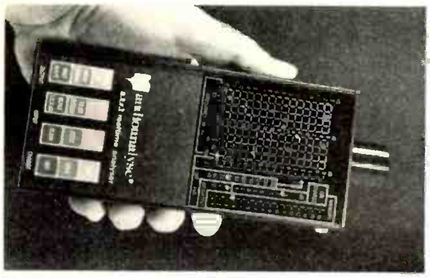
COMMUNITY LIGHT & SOUND
333 EAST FIFTH STREET
CHESTER, PA 19013
(215) 876-3400

For additional information circle #121

NEW PORTABLE REAL-TIME SOUND ANALYZER FROM SPANTA

The Model ATR 1 Real Time Sound Analyzer is described as the first portable unit to incorporate a CMOS memory capable of memorizing 15 acoustical frequency responses, including octave level, SPL level, weighting and gain. The acoustical reading is displayed with bright LEDs next to calibrated scales. A flat frequency response of 20 Hz to 16 kHz assures a high degree of accuracy.

Weighing only 25 ounces, the ATR 1 has a water-tight touch keyboard for easy access to various functions; fast, slow and peak response selection, with peak response having infinite hold time; long-time CMOS memory storage with 1k capacity; scale variation in 1, 2 or 3 dB divisions; display brightness automatically regulated to ambient lighting conditions; blinking LED to provide early warning of low battery condition; and input available for analyzing exterior electrical



signals (microphone, tape machines, etc.), with an output being available for direct recording equipment, or exterior analysis equipment.

SPANTA, INC.
P.O. BOX 193
FRANKLIN LAKES, NJ 07417
(201) 337-0044

For additional information circle #122

INTERFACE ELECTRONICS MODEL 324 COMPACT MIXER

The Model 324 is a 24-track mixer in a compact frame for small recording studios and multitrack location recording. The meter bridge has been eliminated, and each plug-in module contains all the elements of a mixer, including a slider master channel with 10 light LED VU metering, an input channel with track assign pushbuttons, panpot, four equalizers (two tuneable), four cue/effects sends with pre/post fader and module/playback switches, and elaborate solo, monitoring, and playback features.

The basic mixer includes 28 input/output modules, one Type NA monitor module, and an external rack-mount power supply. The unit measures 47 × 29 × 5 inches and weights approximately 60 pounds. The Series 324 may also be ordered with less than a full set of modules (for less than 24 tracks), or with more inputs. Mike inputs are balanced 200 ohms nominal XLR-3 type; modules also have break-in jacks, line inputs, and playback. All connectors are on the rear panel.

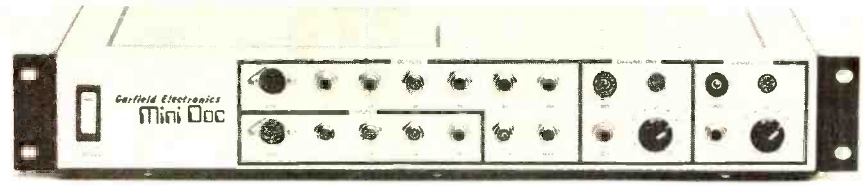
The Model 324 can also be used as a sound system mixer, making up to 24 submixes into a mono or stereo slider house output, plus operator's monitor with solo.

INTERFACE ELECTRONICS
6710 ALDER
HOUSTON, TX 77081
(713) 660-0100

For additional information circle #123

GARFIELD INTROS MINIDOC SYNTH/DRUM MACHINE SYNCHRONIZER

Configured as a low-cost companion to the Doctor Click, the Mini Doc simultaneously coordinates timing for sequencers, drum machines, and arpeggiators from Roland, Oberheim, Sequential Circuits, Linn, Korg, Moog, E-mu Systems, NED



Synclavier, Fairlight, Simmons, Wave PPG, and MXR.

The unit's two independent clock circuits control arpeggiators in 22 synchronized rhythms; all instruments are synchronized to tape. Mini Doc also generates individual triggers from audio. All inputs, outputs, and controls are mounted on the front-panel of the unit, which mounts in a standard 19-inch rack.

Recommended retail price of the Mini Doc is $595.

GARFIELD ELECTRONICS
BOX 1941
BURBANK, CA 91505
(818) 840-8939

For additional information circle #124

New Products

SHURE MODEL SM83-CN LAVALIER MIKE

One problem addressed by Shure engineers in the mike's design is the "chest resonance" phenomenon often encountered in using lavalier microphones. The SM83-CN's wide frequency response has been specially tailored to compensate for this problem, with an electronically created dip at 730 Hz and an acoustically generated HF boost above 3 kHz. The result is said to be an extremely natural sound without boominess or excessive brightness. In addition, the SM83-CN's controlled LF rolloff substantially reduces clothing, handling, and room noise.

These sonic characteristics are made possible through the use of a compact

amplifier supplied with the unit, and which can easily clip onto the user's belt, or fit into a coat pocket. It may be powered by a standard, 9-volt battery, or by Simplex power from an external source.

Four mounting clips also are provided: a single-mount tie-bar; a dual-mount tie-bar (for mounting two microphones simultaneously); and two multi-purpose mounting clips that may be connected to a lanyard or sewn, pinned, or taped onto clothing. Other features of the SM83-CN include: very low noise, minimal RF and magnetic hum susceptibility, a field-replaceable cable that utilizes steel conductors for strength, a dark nonreflective finish, and a foam windscreen for outdoor use.

User net price of the SM83-CN is $210.

SHURE BROTHERS INC.
222 HARTREY AVENUE
EVANSTON, IL 60204
(312) 866-2553

For additional information circle #127

For additional information circle #126

SEQUENTIAL CIRCUITS DRUMTRACKS DIGITAL DRUM MACHINE

Comprising a fully programmable drum machine featuring 13 digitally recorded instruments and MIDI connection, Drumtracks enables volume *and* tuning to be programmed individually for each of its real drum and cymbal sounds. Extensive editing options include auto-correct, overdub, record, erase, and copy modes.

Model 400 Drumtraks features a programmable mixer with a monophonic output (which can drive stereo headphones). For control by external mixers or processors, six audio channels (plus the metronome) are available at the back panel through standard ¼-inch phone jacks. The overall memory capacity of over 3,300 notes can be allocated to up to 100 different drum patterns, any of which can be up to 100 measures long in any time signature. Tempo range is 40 to 250 beats-per-minute. Each overdub of a pattern can be recorded with a different instrument volume or tuning, in real time, or auto-corrected to one of eight levels of resolution. Any part of an instrumental track can be erased; patterns can be copied and added together.

Once drum patterns (sequences) are recorded in the Drumtracks memory, up to 100 songs can be defined. Basically, songs are made by chaining patterns together. Each song can consist of up to 100 steps. Steps specify how the song is built by selecting patterns and inserting volume or tempo changes; songs, too, can be edited, copied and appended.

Two built-in interface systems also are featured: a selectable 24, 48 or 96 pulse-per-quarter note clock input; and a 24-pulse clock output for older sequencers or rhythm units and sync-to-tape. For operation with computer-controlled sequencers, the new MIDI interface is provided.

Suggested retail price of the Model 400

Drumtracks is $1,295.

SEQUENTIAL CIRCUITS, INC.
3051 NORTH FIRST STREET
SAN JOSE, CA 95134
(408) 946-5240

For additional information circle #129

DAUPHIN LINE OF CUTEC MIXER AND EFFECTS UNITS

New models include a 12-channel mixing console, and rack-mountable digital delay, stereo power amplifier, electronic crossover (both two-way and three-way models), and stereo graphic equalizer with spectrum analyzer.

Standard features on each model include: MX 1210 mixer — 12 channels with separate mike/line inputs, overload LED indicators, effect and foldback sends; CD 425 digital delay — hold function for main delay (max 1.024 seconds), reverb effect with combination of main and sub delay, and lo/hi equalizer; PA 902 stereo power amp — 60W per channel into 8 ohms

(90W at 4 ohms, or 160W mono), dual level controls and VU meters, XLR/phone inputs, and banana/phone outputs; CX electronic crossovers — CX 230 features two-way stereo, eight-step dividing points from 250 Hz to 6 kHz, while CX 330 features six-step dividing points between low-mid and mid-high frequencies, and frequency phase switch for each band; and GS 2200 Equalizer —10-band graphic display, real time spectrum analyzer, acoustic level pink noise generator, and input selector switch for line or tape.

DAUPHIN COMPANY
P.O. BOX 5137
SPRINGFIELD, IL 62705
(217) 793-2424

For additional information circle #130

MXR ANNOUNCES NEW 1500 DIGITAL DELAY

The Model 1500 features a full 20 kHz bandwidth at 1.5 second delay, along with a sweep ratio of 10:1. The capabilities on the delay range from a minimum of 100 microseconds to a maximum of 1.5 seconds, with the effects of flange, chorus, double and echo.

The unit, which measures 1¼ inches high by 6¼ inches deep, and is a standard 19-inch rack width, has a suggested retail price of $500.

MXR INNOVATIONS
740 DRIVING PARK AVENUE
ROCHESTER, NY 14626
(716) 254-2910

For additional information circle #131

For additional information circle #132

For additional information circle #128

CREATIVE INTERFACE

— continued from page 32 . . .

mum Volume, Articulation, and Timbre Amount (similar to Filter Amount in an analog system.) Notelist tells the operating system that a playable program sequence follows; "Using 3-4" informs the system that it will play the following sequence of events with the timbre or sound effect stored in memory location 3-4. "/S1" is the MIDI device select command, and tells the computer to which MIDI bus sequence should be sent. (In effect, it selects which synthesizer will be playing the note or sound effect.)

Although this format looks similar to the original Script format, there are some material differences. For example, a number of input formats are available in relative or absolute time values:

minutes:seconds:video frames;
minutes:seconds:film frames;
film feet/frames; and
minutes:seconds:thousandths.

After the timbres are constructed or sampled, and the SFX data typed in, the LinnDrum can be programmed. The only modification that has been made to the LinnDrum, which we use as the master clock for the entire system, is a 10-key keypad for tempo input. (With this modification we can be sure of the LinnDrum's initial tempo, a very important consideration for synchronization to finished film or video.)

When the score is programmed into the LinnDrum, the Apple II+ is re-booted with our MIDI software. The following interface patch is deceptively simple, as can be seen from the accompanying diagrams.

One sync pulse is printed onto tape from the Linn. The click track will be generated live from the Synclavier. If analog-track real estate is not at a premium, timecode and a click are also printed to tape. However, it really isn't necessary to print anything other than LinnDrum sync and SMPTE timecode to tape.

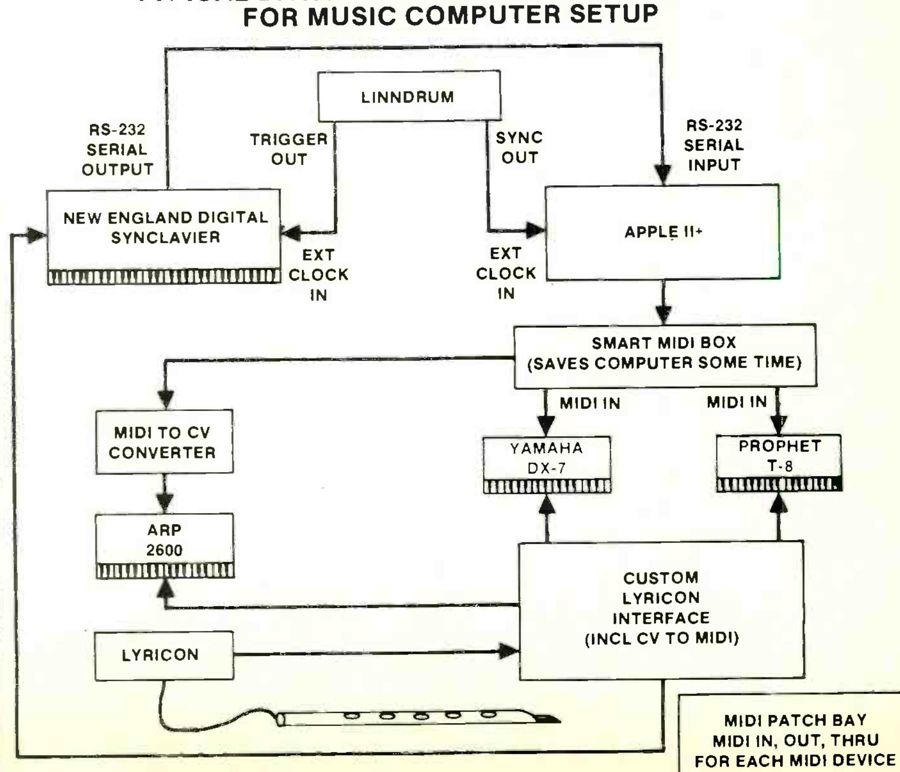
The LinnDrum reads sync from tape, while the Apple II+ reads sync out from the Linn at TTL level. The Synclavier reads the ⅛-note trigger from the LinnDrum, and the Apple drives the T-8 and DX-7. The Synclavier drives its own internal 16-track digital memory recorder and the Apple II+. Everything is 100% in sync, and outputs in stereo.

Obviously, stereo is used for album dates and radio spots. Not so obvious, however, is the use of stereo for differential filtering. Signal processing and differential filtering is used to enhance a signal that will eventually be heard as monophonic. The left channel can be filtered, equalized, and processed separately from the right channel, and then combined into a mono signal. This technique is excellent for creating phantom Doppler effects, and other unique timbre requirements. Since much of our signal processing and all of our stereo panning is done in software, the computer remembers highly complex delay, reverb and EQ settings. As such, it forms an excellent creative tool, and saves hours of mixing time — to say nothing of what it's worth during the inevitable remix.

All of these techniques are leading to a new era of music production; this is only

Master Click Program
The Master Click Program, written by Shelton Leigh Palmer and James D. Kafadar, is currently available for the Apple II, Apple II+ and Apple IIe microcomputers; Synclavier II and IBM PC versions will be released shortly. For further information, contact Shelton Leigh Palmer & Co., Inc. 360 East 57th Street, New York, NY 10022. (212) 980-3445.

TYPICAL DATA AND CONTROL VOLTAGE FLOW FOR MUSIC COMPUTER SETUP



the beginning. Every major company has hardware and software on the drawing board. One of the nicest things about the new technology is that you *don't* have to be a major company to get into it. If you have a soldering iron, a Radio Shack catalog, and a little knowledge of computers, you're on your way.

In future installments of this continuing series of short articles, I'll be considering other roles for computers in recording and music production, including digital sampling and synthesis, plus CV-to-MIDI conversions. ■■■

INDUSTRY INVENTIVENESS IN THE EIGHTIES

Mark Joseph's Bartering for Production Services

by James Riordan

Invariably, necessity is the mother of invention, and the onset of a financial squeeze in the record industry is what provided the necessity for the inventiveness discussed in this column. Some people maintain that this financial squeeze is now over. That may or may not be true, depending on where you sit in our industry, but we can all be grateful that the spirit of inventive conservation is still prevalent throughout the record business. Good times may be on the way back, but most of us intend *not* to lapse into the trap of over-indulgence. After all, isn't that what got us into trouble in the first place?

This month's column deals with an excellent example of not allowing the lack of funds to stand in your way. When Mark Joseph embarked on his current venture into the music business, he barely had the capital to sustain himself. His intention was to launch a first-class production company that only worked with top-flight producers, engineers, and studios. Besides recording new acts, Joseph intended to finance accompanying videos and hire top music-industry professionals to guide these acts into the right deals. If there is one thing that such a list of goals requires, however, it is money. And most of us would stop right there until we had the money necessary to follow through with our plan. That's *not* the way Mark Joseph saw it.

"Instead of looking at what I *didn't* have, I looked at what I *did* have," he recalls. "I had a pretty good relationship with some very knowledgable music people, and I had a lot of experience in a variety of businesses. It was clear to me that I would have to find a way to turn my experience into something of value for these and other music people, and then parlay that into everything I needed for my company. I figured that another thing I had going for me was that sales were off in the record industry, and that meant there would be more opportunity for new people with new ideas, as long as they could make things happen."

Talk about optimism! Joseph knew that some of this nation's biggest fortunes were made during the depression, because the lack of money forced people to be open to new possibilities. The same is true today. Any positive thinking book will tell you that there is a positive seed in every negative situation — all you have to do is find it.

Mark Joseph had been used to finding the positive seed ever since he was involved in a crippling auto accident some years ago; he had broken his neck, and was partially paralyzed. The doctors told him there was only a two-percent chance that he would ever walk again. Six months later, after extensive exercise therapy during which he had to "reprogram the spinal cord," Joseph walked out of the hospital. He has been a positive thinker ever since.

Much of Joseph's business background had been in retail and wholesale floor covering. It would seem that this kind of experience would have very little to do with getting ahead in the music business, but there are no boundaries when creative thinking is applied.

"I found that I could join a barter exchange where I would provide floor covering and installation in return for a combination of cash and barter credits on the exchange," he explains. "The credits were

. . . continued overleaf —

good for a variety of goods and services from other companies on the barter exchange. I then built up my barter credits until I had amassed a fairly significant amount. I went to people I respected in the music business, and worked out a structure for my production company that revolved around several different music industry pros. Each of them is an expert in a different aspect of the music industry, and they all do work for me in exchange for credits on the barter exchange."

Well, why not? Music industry pros spend money on carpeting, clothes, automobile repairs, restaurants, furniture, plumbing, tires, accounting, dentists, art, bicycles and more — all of which is available on most barter exchanges. Joseph then expanded the floor-covering part of his business by allying with noted interior designers Charles Bruno & Associates. Bruno has quite an impressive list of clients, and Joseph is able to provide the "best floor coverings at the best price." Before long, Joesph had turned this new alliance into another plus on the musical side of his company by negotiating interior remodeling agreements with many top recording studios.

"Charley Bruno was the key we needed to expand our floor-covering operation into an interior design company which would have more overall appeal for direct barter. When we can directly barter with a company we aren't required to use the exchange. This makes it more profitable for us, and allows us to give the customer an even better deal. Of course, not all of our work is on a barter basis. We sell a great deal of floor covering now and, in conjunction with Charley, assist him in some very large interior design projects. The barter system is now used to fill in the gaps. When I can't get cash business, I can always get something happening through the barter system. The result is that our crews are working all the time, and we have to constantly expand them.

"Of course, I enjoy the work we do in the music industry the most because it invariably benefits my production company. I am very proud of the work we've done with the Robb family over at Cherokee Studios [Los Angeles], and with Buddy Brundo at Conway Studios [Hollywood]. These are two of the finest recording facilities in our industry, and anything that is done for them is noticed throughout the industry."

The structure of Better Days, Joseph's company, may appear a little strange to most businessmen — having a music division and an interior-design division — but there is no question that it gets the job done. Besides the full construction and floor covering crews in the interior design division, Better Days has begun projects in recording, music video, and film. The company has its own full-time accountant, and a top West Coast attorney. The company's expanded 12-man board of directors include an interior designer, building contractor, film producer, nightclub owner, concert promoter, session musician, record producer, music-video director, personal manager, plus the accountant and attorney. They have been involved with the careers of such noted artists as Stevie Wonder, America, Michael Jackson, Dionne Warwick, Three Dog Night, The Eagles, Earth, Wind, & Fire, Alice Cooper, The Commodores, Steve Miller, and many others.

"I feel the key to our company is the quality of people we have on our board," Joseph enthuses. "I didn't just go out and grab the first people we could get. I felt I had something of real value to offer them and I wanted more than just a hired gun, so to speak. All of the people on our board have a good deal of integrity, and they all believe in what we are doing."

Today, Better Days is well into the black with a strong cash flow, and a vast amount of barter power. The company has made strong allies in the music industry, and hopes to complete recording on an album and corresponding music video in the early part of the year. Mark Joseph has proved that talent and hard work can be turned into profit, no matter what the current economic conditions.

"There are two ways to look at an obstacle," he concludes. "You can see it as something that will stop you, or something that will spur you on to do an even better job at what you are trying to do. Obstacles force people to be more creative and work harder, and sometimes that will produce better results than you ever anticipated. In our case, it has done exactly that."

Joseph is currently at work on a book about positive attitude, and overcoming obstacles through faith and hard work. "I don't know when I'll ever have time to complete it but I keep thinking more and more about it. For me, that's the first step. Once I've got it decided in my mind I know I can make it happen." ■■■

news

— continued from page 29 . . .

INTERNATIONAL MIDI ASSOCIATION

The International MIDI Association's purpose is to keep users informed with up-to-the-minute, accurate information regarding the rapidly evolving world of MIDI. The IMA also offers to its members a database service.

The Association's newsletter, *IMA Bulletin*, is described as the first publication dedicated totally to MIDI related products and news.

Further details are available from Roger Clay, IMA, 8426 Vine Valley Drive, Sun Valley, CA 91352. (818) 768-7448.

VILLAGE RECORDER AND MRI MERGE

Village Recorder of Santa Monica, CA, and Motionpicture Recording, Inc., one of Hollywood's newest film audio post-production companies, have joined forces to create what is described one of the largest audio facilities outside of any major studio lot in Los Angeles.

The new post-production entity — to be known as Villge/MRI — will offer three, 24-track studios, video off-line editing, a 35mm four-track stereo dubbing theatre, a film and video ADR stage, a film Foley stage, a Dolby Stereo optical transfer facility, editing suites, and production offices.

Both companies will remain autonomous in management, but will be working collectively in furnishing film producers with complete production packaging for audio pre- and post-production. Village Recorder's Michael Geller will supervise and manage the Santa Monica division of Village/MRI, while Garry Ulmer, owner and chief engineer of MRI, will continue to head up the film facilities of Hollywood's Village/MRI.

AUDIO KINETICS RE-ORGANIZES U.S. OPERATION

Contrary to industry rumor, Audio Kinetics, manufacturer of the Q.Lock series of SMPTE synchronizers, is expanding its U.S. operations. The England-based manufacturing company now wholly owns its American subsidiary, located in Los Angeles, and plans are well under way to set up a nationwide network of specialist dealers and service agents, both for Q.Lock and the recently introduced MasterMix disk-based console automation systems.

John Frazer, VP of the North Hollywood office, states that, "We aim to provide more support to both customers and dealers, having now established the product throughout North America." In addition, technical manager Sean Fernback will be providing customer and dealer support. Frazer and Fernback can be reached on: (818) 980-5717 for sales and service information. The company's address remains unchanged at: 4721 Laurel Canyon Boulevard, Suite 209, North Hollywood, CA 91607. Its toll-free number is: 1-800-423-3666 (outside California).

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news

DIGITAL MUSIC SYSTEMS HOSTING DIGITAL SEMINAR

The two-day workshop, to be given on April 16 and 17 in Boston, will cover the basics of digital audio and its application to the recording and synthesis of music.

Sampling, quantization, and other digital audio basics will be presented from a non-technical point of view; participants will be assumed to have no background in digital audio or electronics. Digital recording, the Compact Disc, digital signal processors, and digital synthesizers will be among the applications discussed.

The course will be taught by Dean Wallraff, a former member of the faculty at MIT's Computer Music Studio, and the founder of Digital Music Systems, Inc.

The course fee is $150, and the registration deadline is March 9. For more information contact: Digital Music Systems, Inc., P.O. Box 1632, Boston, MA 02105. (617) 542-3042.

ZENITH AND DBX SELECTED FOR MULTICHANNEL TV SOUND TRANSMISSION SYSTEM

The Electronic Industries Association's Multichannel Sound Subcommittee has recommended adoption of Zenith's transmission system and dbx noise-reduction system for multichannel television sound. The industry's unified position was filed with the FCC on January 30, 1984. And in a simultaneous announcement, the National Association of Broadcasters has asked the Federal Communications Commission to adopt technical standards for a *single* multichannel television sound (MTS) system, and not rely on the marketplace for development of the expected new services. It says that a single system would result in the "swift introduction of high quality, low-cost stereophonic sound and separate audio program services." In noting the EIA's endorsement of the Zenith/dbx system, the NAB offers that such a move will give the Commission a solid basis on which it can review, select, and adopt the recommended system.

The system selected by the EIA Committee permits the transmission of stereophonic sound programming, and is compatible with existing television receivers. The system also contains a separate audio program channel that can be used for foreign language, educational, or other purposes.

In essence, the Zenith transmission system broadcasts a left+right signal on one channel, and a left-right on the other; to maintain compatibility between mono and stereo domestic television receivers, only the L–R channel is compressed by dbx noise-reduction, for subsequent expansion/decoding at the receiver.

According to Les Tyler, VP engineering at dbx, the noise reduction encoder performs five signal-conditioning function: 1) lowpass filtering: to remove out-of-band signals; 2) wideband compression: to keep overall signal levels well above transmission channel noise while avoiding overmodulation; 3) spectral compression: to maintain proper spectral balance in the transmission channel; 4) static pre-emphasis: to match overall spectral requirements imposed by the channel characteristics; and 5) pre-emphasized clipping: to eliminate peak overshoots.

At low and midfrequencies (100 to 3 kHz), where channel dynamic range is widest and where most of the primary signal energy resides, the effective compression ratio is kept to 2:1, with relatively slow time constants, to maintain transparency of operation. At high frequencies, where masking of noise components is especially critical, the effective compression ratio rises to 3:1, with faster time constants. Passive R/C networks limit the maximum amount of dynamic pre-emphasis to optimize the amount of audio processing at high frequencies.

— News Notes —

Soundout Laboratories' products are now being distributed in the U.S. by **MCI, Inc.**, of Waco, TX. Heading up the new team at MCI is new product manager **Scott Pelking**; further details from MCI, Inc., Box 8053, Waco, TX 76714. (817) 772-4450 ... **Joiner-Pelton-Rose, Inc.** the Dallas-based acoustical consulting firm, has moved into a new 9,200-square-foot office facility. According to **David Joiner**, JPR president, the firm maintains a permanent full-time staff of over 20 individuals whose only concern is that of acoustics-related studio designs. The company's new address is: 4125 Centurion Way, Dallas, TX 75234. (214) 392-7800 . . . **Kurzweil Music Systems, Inc.**, manufacturer of the Kurzweil 250 digital keyboard (see December issue, page 123), has relocated to expanded quarters in Waltham, MA, which includes 34,000 square feet of office, manufacturing, and warehousing space, almost four times the size of the company's previous offices. The new location is 411 Waverley Oaks Road, Waltham, MA 02154. (617) 893-5900 . . . **Digital Entertainment Corporation,** the U.S. affiliate of Mitsubishi Electric Corporation, and responsible for all professional digital audio systems marketing in North America, has opened a 2,500-square-foot New York City sales and support facility. At the formal opening, DEC president Tore Nordahl stated: "The expansion of our U.S. operations into the heart of the New York entertainment studio neighborhood confirms Mitsubishi Electric's total commitment to professional audio, in both music recording and broadcasting." The new office address is: 555 W. 57th Street, Suite 1530, New York, NY 10019. (212) 581-6100 . . . **Eventide, Inc.** has moved to a new 20,000-square-foot facility at One Alsan Way, Little Ferry, NJ 07643. (201) 641-1200. In addition to increased production facilities for existing product lines, the new plant has enabled the company to step up development and introduction of new products . . . **Trident (USA), Inc.** has appointed **Audio-techniques, Inc.**, to be dealers for the com-

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Gordon Hardy, President & Dean • Jorge Mester, Music Director

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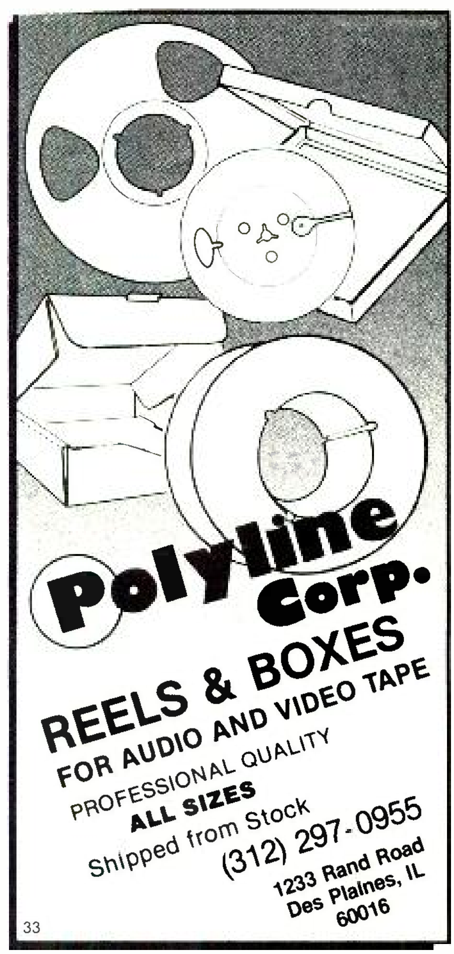
For further information write:
The Aspen Audio Recording Institute
The Aspen Music Festival
1860 Broadway Suite 401
New York, New York 10023

The Aspen Music School admits students of any race color, national or ethnic origin

For additional information circle #141



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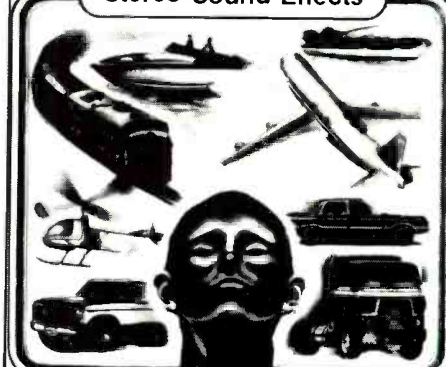

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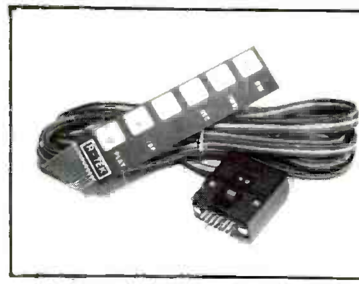

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news

— continued from page 159 . . .

pany's entire line of consoles and Optimix computerized console automation. Also, Trident (USA) and **Wilson Audio Sales** have named **Trackside Engineering** as the exclusive Georgia dealer for the firm's range of audio consoles. Trackside is nearing completion of its own in-house "mix room" which, when completed, will function as both a multitrack mixdown room and an equipment demonstration suite. Trackside wil continue providing full service and support in the southeast, while Wilson Audio Sales, as manufacturer's representative, will provide assistance to Trackside, who can be reached at 2670 South Cobs Drive, Smyrna, GA 80080. (404) 436-3024 . . . **Tannoy** has begun a major marketing effort in the U.S. and Canada, with the establishment of a new factory-backed operation in Ontario. The facilities include offices, warehousing for products and parts inventory, plus complete sales and service functions. **Wib Heuchroth**, managing director, and **Bill Calma**, marketing director are based at the new offices; 97 Victoria Street North, Kitchener, Ontario, Canada N2H 5C1. (519) 745-1158 . . . For the second consecutive year, **Dolby Laboratories, Inc.** will sponsor the Sound Achievement Award of the annual Nissan FOCUS Awards Competition. FOCUS (Films Of College and University Students) is open to 16mm film works produced on a non-commercial basis in conjunction with an American educational institution. The Dolby-sponsored Award carries a $1,000 cash prize, and a trip to Los Angeles, where all FOCUS winners are treated to special tours and film screenings, and a series of informal seminars on filmmaking. □□□

— People on the Move —

• **Chris Foreman** has joined the Audio Systems Division of Peirce-Phelps, Philadelphia, PA, where he will develop marketing of the various professional products and serveces offered by the division. Foreman has held previous positions with Stanal Sound, Gary Davis and Associates, Altec and, most recently, Community Light & Sound.

• **P. Woody Jackson** has been named national sales manager of Klipsch & Associates, from his former position as western regional sales manager.

• **David G. Kennedy** has been appointed the new president of dbx, Inc. "My primary objective as president of dbx," Kennedy says, "will be to maintain the company as a technology leader, while directing the company's growth and expansion through the electronics age." Kennedy, former VP of finance for Instrumentation Laboratory, Inc., served with that company for over 17 years. From 1967 to 1977, he also served as director of Info Inc., a diversified consumer durable products company.

• **Barbara Ann Adler** has been promoted to the position of regional distribution manager for Agfa-Gevaert's Dallas regional distribution center, which supplies the southwest region of the U.S.

• **Andrew Schatz,** who joined Everything Audio in 1981, has been promoted to chief audio engineer for the company. Schatz previously worked as an independent engineer, and as a sales consultant for K & L Pro Audio in Boston. Also, **Lon Le Master** recently joined the company's sales force from Neotek West. His industry experience includes working for Val Garay at Record One Studios, where he assisted in the engineering of the *Toto IV* album

Performer, composer, and producer Frank Zappa, one of the first artists to acquire a complete digital recording system, recently played host to several Sony executives at his Los Angeles studio, which includes a Sony PCM-3324 digital multitrack and PCM-1610 two-track digital mastering unit for preparation of Compact Disc master tapes. Seen from left:Dr. Toshi Doi, deputy general manager, Sony Audio Products Group, Zappa, and Curtis Chan, Sony U.S. digital audio engineering manager.

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