Directory: Consoles and Mixers

MODERN RECORDING & MUSIC

OCTOBER 1985 VOL. 11 NO. 10 $2.25

TENTH ANNIVERSARY ISSUE

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Producer Steve Levine

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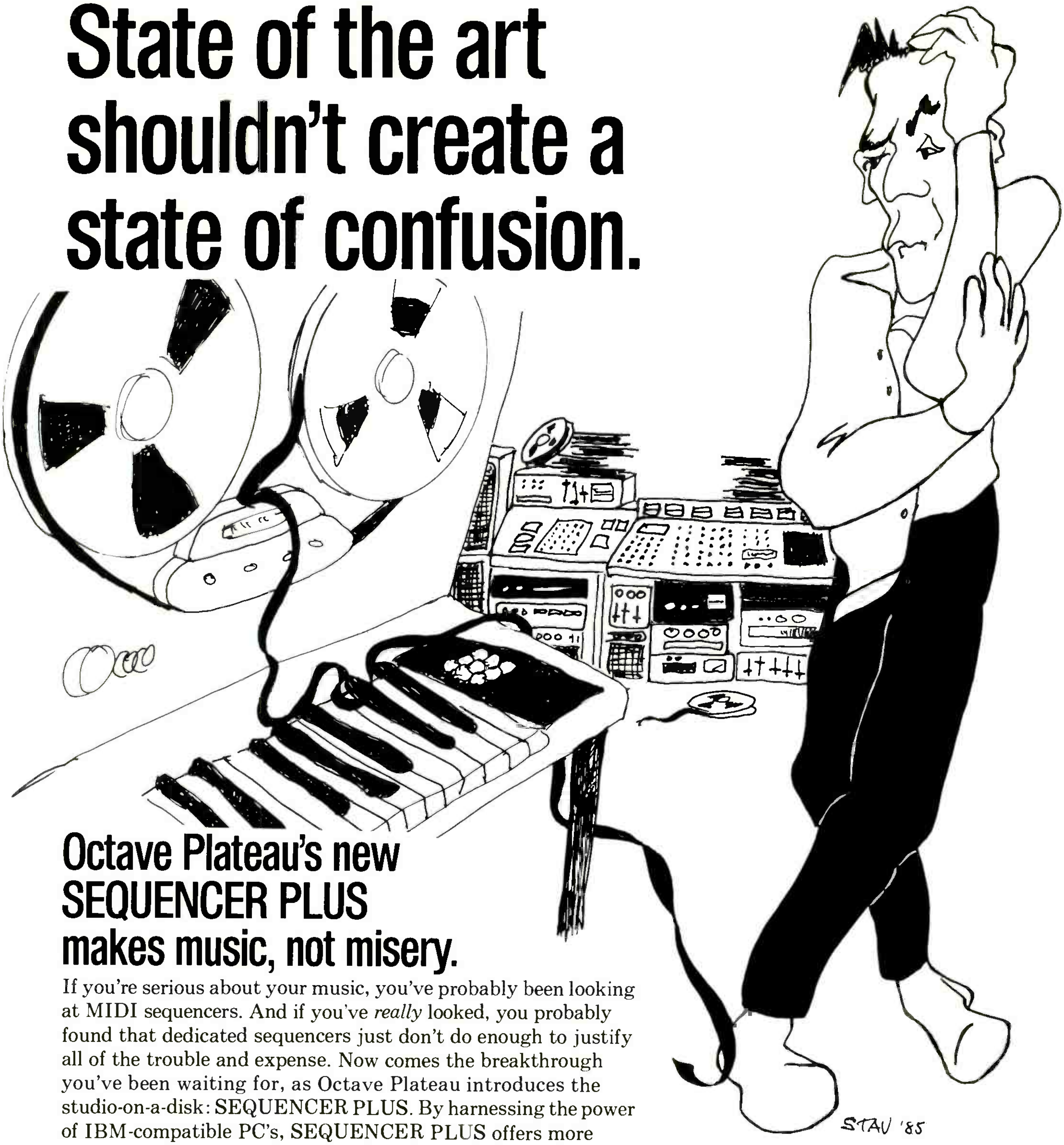

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OCTOBER 1985
VOL. 11 NO. 10

MODERN RECORDING & MUSIC

FEATURES

SOUND IDEAS

Cover and spread photo courtesy of Harry Langdon Photography.

DEPARTMENTS

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MODERN RECORDING & MUSIC

Letters

To The Editor

We would like to thank you for including *Rightersound 1* in your directory "Modern Recording Updates Music Software." The response has been tremendous! As a service to your readers who are DX-7 owners, we would like to add that a demo tape is now available for $10.00; this includes a $5.00 rebate coupon towards the purchase of *Rightersound 1. Notable Software* is proud to bring realistic sounds to DX-7 owners at a reasonable price, and thanks to your readers for their continued support.

—Dennis Righter
Music Specialist
Notable Software

4-Tracker Comments

I read and enjoyed Bob Buontempo's recent article *"Poor Recorder's Almanac: Part 1."* I have been a reader of *MR&M* for a number of years, and have found it very informative.

In Bob's article he mentioned that he would welcome some comment about what the reader would find helpful, giving an example of a "Soundsheet" or cassette. I would welcome such a help as I'm sure it would be very effective in demonstrating the concepts you are presenting.

In his article he also mentioned that a number of projects were outlined as construction projects in *MR&M*. It would be very helpful to the new reader, as well as those of us who have missed these articles, if these "How To" articles could be compiled into some type of book or pamphlet form and offered for sale to readers. I am aware that back issues containing these articles may be available, but the cost of obtaining all the nècessary issues could add up to quite a bill. If Xerox copies could be made of the construction articles and offered for sale, possibly in kit form with the aforementioned cassette, I'm sure you'd have some takers.

Once again I thank Bob for his interest in us "Little Four Trackers" and hope he will find some wisdom in my suggestions. Keep up the good work, I eagerly await the next part in the series.

—G. Allan Clarke
Stormwatch Productions
Berrien Springs, MI

Thank you for your kind words. We hope you continue to find MR&M *informative and helpful.*

The possibilities of the Soundsheet Bob mentioned is currently being investigated by us here at MR&M. *If you'd like to see a Soundsheet in the magazine, please drop us a letter and let us know. As far as the construction projects go, this month (Oct., our 10th anniversary) we are running the projects Bob mentions —and more. So stay tuned! Any letters should be addressed to:*

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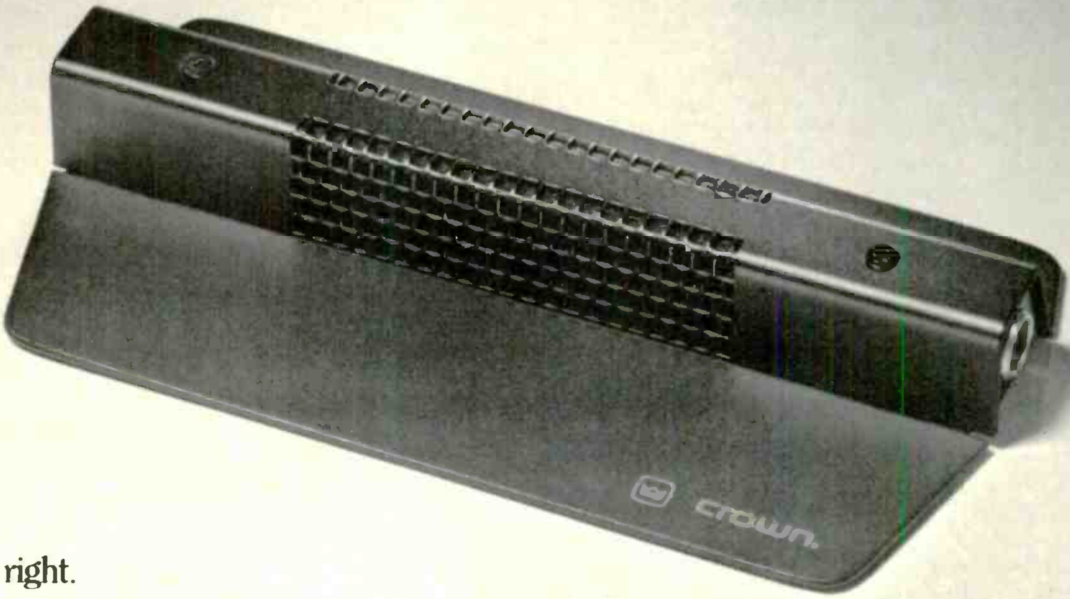

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Talk Back

For this month's 10th anniversary issue, we went back over the past few years to find several Talkback letters that answer some age-old questions.

Reprinted from Feb. 1983.

Demagnetization

Like many people these days, I have learned about recording, not by being apprenticed to a professional engineer, but by reading and experimenting with my home studio. As a result, I have lots of questions and nobody to turn to for answers, especially when it comes to the arcane and mysterious process of demagnetizing. I have never seen anybody else demagnetize a tape head, and the written instructions I've encountered are far from comprehensive. I could even be doing something horribly wrong that will destroy my tape heads. So here are a few of the things that have been puzzling me.

1. When moving the head of the demagnetizer back and forth over the tape head (well, actually, under the tape head) should I move only in a direction parallel to the head gap, that is, perpendicular to the face plate of the tape head, or should I move from side to side, in a circular manner, or in some other pattern?

2. Should I continue moving the demagnetizer in this "scrubbing" pattern as I draw it slowly away from the head, as one of my friends recommends, or should I stop scrubbing and draw it away in a basically perpendicular line?

3. Will it harm the tape head if the metal tip of my cheapo demagnetizer accidentally comes in contact with the head itself?

4. How close should the scrubbing motion be done, anyway? I have been doing it about ⅛-in. from the head.

5. Does the tip of the demagnetizer have to be square to the orientation of the head, or is it okay to be angled slightly to the left owing to the fact that I'm left-handed?

6. I know I'm supposed to back the demagnetizer off at least three feet from the deck before turning it off. But do I have to back it away this far between demagnetizing the record head and demagnetizing the playback head, or can I just back it off a foot or so, so that the effective distance is roughly the same to each head?

7. After I've demagnetized one head and am bringing the demagnetizer in to demagnetize another, do I have to approach as slowly as I drew back, or can I come in faster? How much faster?

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8. Do I have to demagnetize the little metal posts that the tape travels over between heads? If so, should I demagnetize them before or after I demagnetize the heads? Won't the magnetic field coming out of the side of the demagnetizer's tip have some effect on a nearby head even when I'm demagnetizing one of the adjacent posts?

9. Do I have to rotate the tape guides on the spring-loaded tension arms and demagnetize them from all sides, or will just demagnetizing them from one side demagnetize the whole roller clear through?

10. How often do I have to go through this whole ritual, anyway?

—Jim Aiken
Cupertino, CA

O.K. Jim, let's start with your last question. There is only *one* way to determine the need for demagnetization and that's with a magnetometer. Now before you faint, let me say that this device is both inexpensive and simple to operate. It's like a VU meter except its zero is dead center. An excellent version is made by R.B. Annis of Indianapolis. You just touch its bottom edge (or its clip-on probe) to the suspect part and it'll read negative (left) or positive (right) magnetism in "gauss." A scale of no more than five gauss is necessary and as an explanatory note the earth's magnetism is about ½ a gauss.

In the studios I work out of, the machines are checked daily. This is due to the high volume of work done, as just running tapes can build up magnetic fields. Additional causes of magnetization are things like speakers or transistor radios placed near the deck. Occasionally a meter or even the earth's field can cause problems. Recording tape is very sensitive to extraneous magnetism. Exposure can degrade the signal by limiting or decreasing its high frequency content as well as increasing background noise (hiss) several dB. In other words, your signal to noise ratio goes down the tubes.

Your heads are made up of "soft" metal. They are very permeable and therefore magnetize very easily. However, once the influence of a magnetic force is removed, its retention or memory of that source is slight. Magnetism can build up, but demagnetization is easily accomplished. *All* the metal parts along the tape's path such as capstans, guides, rollers and springs can not only pick up magnetism but pass this on to the tape. These are "hard" metals and their retention is such that they can actually become magnets. If your "cheapo" demagnetizer is rated at less than 300 Oersteds it wouldn't be of much use here. What's an Oersted? It's simply a measurement of magnetic intensity and should be listed on your instrument's specification sheet.

The way that demagnetizers work is easily understood and understanding is the key to knowledgeable use. When you use a demagnetizer, you're exposing a metal part to a magnetic field of reversing polarity (neutral) such as 50 or 60 cycle alternating current circuits, then reducing this intensity gradually by pulling the magnetic field away. This is called cyclical demagnetization and the actual demagnetization occurs *only* during the reduction of intensity (pulling away period).

When starting demagnetization the parts should be rotated or the demagnetizer waved to insure optimum neutralization. You should

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always pull away to at least one to two feet for each part demagnetized. Follow these simple guidelines and you'll never have to worry about going through this whole ritual again.

1. To avoid occurrence of accidents do not keep demagnetizer in control room or anywhere else tapes are stored.
2. After cleaning deck, check parts with magnetometer.
3. Check to insure no tapes are around the area before plugging in demagnetizer.
4. Turn on at least one to two feet away from deck.
5. Approach to ⅛-inch (do not touch heads).
6. Wave slightly sideways or rotate part.
7. Don't hold in that position, but slowly pull away at no more than three to four inches per second to a point at least one to two feet away.
8. Never interrupt the demagnetizer's power during this cycle.
9. Continue demagnetizing where needed following the same steps.
10. Check all parts again with magnetometer.
11. To clear any magnetic particles out of the head gap, soak a lintless tissue in head cleaning fluid (less than 10% water) and stick this to head. Now when you approach the head, move the demagnetizer once down and once up the length of the gap, then demagnetize as usual. Any particles will adhere to the tissue.

In any case, once you have a magnetometer, you'll be able to answer your own questions about the adequacy of demagnetizing methods.

—Mike Shea

Reprinted from Feb. 1982.

Phase Maze

I hope you can help explain a situation that has me rather confused. Dealing with 3-pin microphone cable for balanced and unbalanced line sends, as well as microphone signals, I have been unable to determine a uniform pin assignment. Pin 1 is the shield, but pins 2 and 3 seem to alternate as "hot" and "cold" (in-phase and out-of-phase, respectively). Pin 2 is usually "hot" on microphones, but on balanced lines, "hot" can be pin 3 (witness Peavey equipment) *or* pin 2 (as on Biamp units), or *either* as is the case with Yamaha's unbalanced line inputs.

I am confused by this seeming lack of uniformity and I hope there might be a standard I am not aware of that will simplify things for me.

—Doug Klug
Wausau, Wisc.

There is indeed a lack of standardization in the pin numbering of XLR-style connectors. Fortunately, the majority of the known universe recognizes pin number 1 as the shield connection. However, there are two possible arrangements for pins 2 and 3 in terms of "hot" or "not."

From what I currently observe, mic manufacturers almost always call pin number 2 "in-phase" and pin number 3 "out-of-phase." On the other hand, makers of electronic equipment that interfaces in line level environments (tape recorders, mixer outputs, electronic crossovers, equalizers, limiters, etc.) tend to use pin 3 for the in-phase connection.

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And to further complicate matters, some manufacturers of mixers wire mic inputs with pin 3 "in-phase." This goes against the grain of the standards set by most mic makers.

So what's the bottom line? Simply this: relative phase is much more important than absolute phase relationships. This means that all mic cables and inputs within a given system *must* use a consistent pin numbering system. Consequently, all mics will be in phase with each other.

Put another way, any cable connected to pin 2 on one end must connect to pin 2 on the other (and the same goes for pin 3, of course).

As long as a consistent wiring scheme is followed, out-of-phase cancellations and the resulting audio colorations won't be a problem. This goes for line level equipment as well.

Some authorities have determined that the human ear can detect absolute phase relationships. For instance, if a snare is mic'ed through an audio system, the ideal action of the loudspeaker's cone would be an outward movement (toward the listener) when the drum is "whacked." If the phase is reversed somewhere along the line, the cone will "suck in," which reportedly results in less realistic reproduction.

In my observations, this problem is relatively minor, particularly in typical club and P.A. systems. It is much more important for all mics/processing electronics/speakers to be properly phased with respect to one another.

If you do desire to maintain consistency throughout a sound system, it will be necessary to determine "which pin is which" in each component. This will require some homework: Study manufacturer's literature carefully. Then, pick *your* standard and rewire any equipment that is not in conformity.

One last thing, a 2-, 3-, or 4-way speaker system has its own phasing problems due to phase shifting effects of the average crossover network as well as the physical placement of the one speaker's element versus the other's. So, absolute phasing accuracy can go out the window in this situation. Just remember, though, that the relative phasing of each part of the system is by far the most important.

Be logical and consistent and you won't have any sonic problems slap you in the phase (sorry!).

—Brian Roth

Reprinted from Nov. 1984

Absorbing Sound and Costs

I would like to know the most effective and/or least expensive way to build a soundproof room or booth. Also, I've come across the term Sonex on occasion in your magazine. Can you explain what it's used for, as well as where I can purchase it?

—Floyd B. White
Augusta, ME

We received the following reply from C. Nicholas Colleran, president of Alpha Audio.

Most effective and least expensive can often be conflicting terms. Relative expense can vary with your point of view and income status. There also appears to be much confusion as to the meaning of the words "sound proof."

There are basically two types of sound control materials. These are barriers and absorbers. Sonex, the material you mention, which we distribute, is a sound absorber. Its

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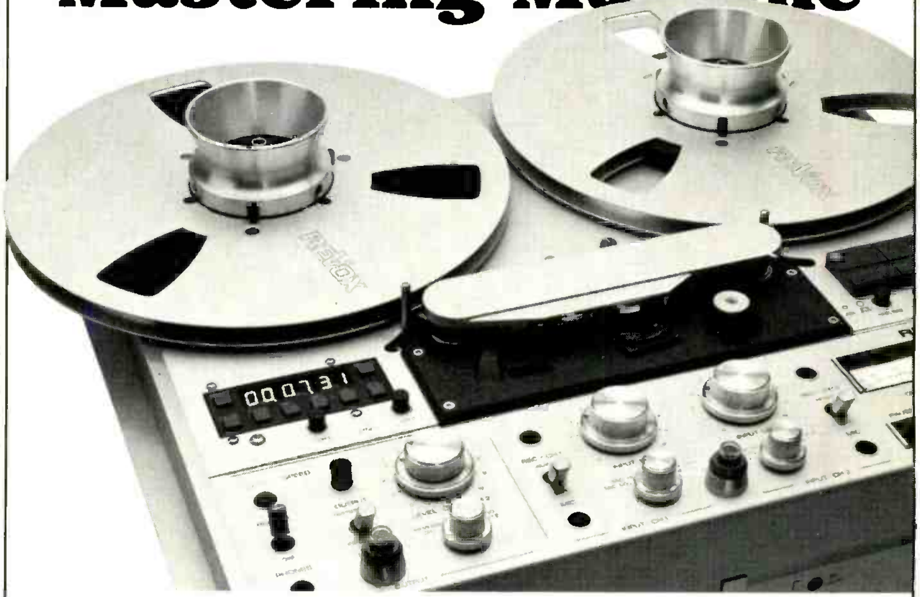

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patented anechoic wedge shape increases the absorption surface 4½ times and yields a 1.00 sound absorption coefficient above 500 Hz. (100% absorption). While the elimination of reflections significantly lowers what is perceived as the sound level in the room, the absorber does little to prevent transmission of sound from one room to the outside or from the outside into the recording room. The actual transmission loss is about 14 dB with Sonex.

Sound transmission loss is accomplished by dense, massive materials and decoupling of the barrier walls. The most massive and compact material we know of is Acoustilead. This can be applied to an existing wall and increases the thickness by only a fraction of an inch. Not only will the lead seal the acoustic paths for air-borne sound, it will lower the resonant frequency of the wall to reduce transmission through the structure.

Both of these materials may appear expensive, but significantly reduce the labor cost of accomplishing your purpose. If you have a lot of time and a little money, you can achieve isolation and high transmission loss by using multiple overlapping layers of sheet rock with the joints well sealed. The inner and outer walls should have separate stud work. The studs should alternate and the gap should be filled with an absorber such as fiberglass.

To the best of our knowledge, there is no economical way to achieve the full effect of Sonex with an alternate material. Good absorption characteristics can be accomplished with fiberglass-lined cavities faced with peg board or other flexible material mounted so that the depth varies from one end of the wall to the other. When using fiberglass, care should be taken that it is covered with an acoutically transparent material which will prevent glass particles from entering the air you breathe. Such a material is available as Sound-tex, a wall covering which has a .20 noise reduction coefficient as well.

Foolhardy Footswitching

First let me commend you on your fine and truly informative publication. I have been receiving *MR&M* for eight years now and I have accumulated a very comprehensive library on equipment, recording techniques, and set up. I have also catalogued a dozen or so construction articles from the pages of past issues. Although I haven't built all of them (though I hope to one day) the projects I have completed have been educational and invaluable tools in my home studio. I am writing in regard to the article by (my hero) Craig Anderton—"Footswitching Your TEAC 3340." My question is this: what pins need to be connected to put my machine into play with an additional switch? Without the third arm Craig refers to in the article, I have to put my recorder into play well before the punch-in point so I can get situated for the part to be dubbed. If I had the play feature it would make an already invaluable tool even better. Keep the articles coming Craig. I look forward to each issue.

—Steve Bucher
Akron, Ohio

Thanks, Steve, for all the kind words. We passed your letter on to Craig and as usual he saves the day.

It's simple to footswitch the PLAY function: Connect one terminal of an SPST normally open push button footswitch to terminal 5 on the 3340's remote plug, and the other switch terminal to remote plug terminal 7. This connection parallels the machine's internal PLAY button. Good luck with the mod!

The Microphone Handbook

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Innovative Crown technology shaped by advice from Showco has produced a more powerful, lighter and smaller amplifier ideal for the touring professional. Higher power, less weight and less rack space translate into critically needed efficiency on the road.

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The performance of the new MT-1000 further strengthens the dependability of Crown amplifiers. The only reason Showco has had to touch their new amps is to move them. And move they have, 78 shows in 52 cities without a failure of any kind. Not surprising for a Crown product but unheard of in any product fresh from the assembly line. Months of Showco's heavy duty field testing has established the MT-1000's reliability even before it hit the market.

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by susan borey and mark oppat

Sound Advice

The House Engineer— Part 2

This month we conclude our discussion with Danny Kapilian, audio engineer, on the responsibilities of the house sound engineer. Steadily employed by a club or theatre, the house, or staff soundperson's job extends beyond the mechanics of mixing. Danny sees the responsibilities as being three-fold: keeping the venue's equipment in good running order, mixing shows for artists who do not bring in their own soundperson, and providing assistance to guest engineers who come in with artists performing at the venue. Last month we covered the first two areas, and now we resume with the third.

Danny Kapilian: When a band comes to your club without their own engineer, you, the house engineer, should deal directly with the artist regarding basic questions about equipment placement and instrumentation. If the band brings in their own mixer, he or she should consult with you about these things. The band's mixer will usually supply you with a stage plot of the physical layout of the band and a mic chart with the input list of which microphones they prefer. Hopefully, egos will not clash. There are enough egos on stage performing, you know?

If the guest mixer is smart, he or she will leave a lot of the set-up, sound check, basic mic'ing levels, and equalization, up to the house mixer before he or she steps in to fine tune everything. For one thing, you're feeding their egos if you let them get things started on their own. You're also doing yourself a favor because they're the ones who know their own house and system the best. They know the limitations, the dead spots, etc. The better house engineers will bend over backwards to help you if you come in with the right attitude and give them all the breathing space available to let them show off a little bit. They want to show you what their club and system are capable of, how well they've maintained it, and how well they've tailored it to the shape and size of the hall. After you've allowed them to "set the table" for you, you can come in and adjust things such as mic placement, particularly on drums.

Another reason the house soundperson will begin setting the levels and basic equalization on the board

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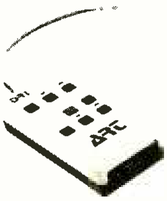

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during the soundcheck before the guest mixer steps in is so that they can get a feel for the band's stage levels and the kind of sound they're putting out. This will enable them to shape the system around that particular act that night. The guest mixer can then step in and adjust levels, effects, and equalization exactly how he or she is used to having them. You can also save time by being diplomatic with the house soundperson; they can probably set up much faster than you could because they do it night after night.

Modern Recording & Music: How do you learn how to step back gracefully and let someone else at your board?

DK: It's hard for some engineers who are on tour with a band and have a big ego. They feel that they're the only ones who should work with the band. It's something that is usually gained through experience. You begin to lose your ego and get into a sharing and networking of talents. It's something house engineers should want to achieve in those circumstances, as well.

MR&M: During the show, what can the house engineer do while the guest engineer is mixing?

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DK: There are some important responsibilities. At the very beginning of the show, you should make sure that the band's mixer has things under control enough so that they aren't going to blow any speakers or send the board's VU meters unnecessarily in the red. Just make sure they're more than competent enough to handle what is essentially your gear. If you see that things are well in hand, and that they are smart enough to achieve the level and tonality they want without jeopardizing the dB level at any end of the equalization spectrum, then you can free yourself to walk around the club a little on behalf of the band's engineer to discover dead spots or other problems in the room. If the club has several levels, you can go make sure there's a proper balance being sent everywhere. You can walk around and be the ears away from the mix for the guest mixer. You should also just be around, if necessary, to answer any questions they might have regarding patching, use of any house outboard gear, or in case something goes wrong. When everything is working fine, it might not be necessary for you to hang out and be there all the time for them. Some house engineers go hang out in their tech office and ask for someone to send for them immediately if they're needed. Sometimes they can keep an eye on things from there or there'll be a headset connecting that room with the board, so they can stay in touch.

* * *

This month, we also have room to address one question sent in by Gary Gray of Stafford Springs, Connecticut. He writes:

"...in the past I have had problems with hum from electric guitars. The problem appears to be that the pickup(s) need to be grounded. When I run a line from the pickup to a ground (such as a cold water pipe or the screw on an electrical outlet), the hum disappears. Is this a safe way to do this? Any suggestions?"

We were wondering, Gary, if you've connected a ground wire from a player's problematic guitar to a pipe or outlet only while checking the problem out, or if you get rid of buzz this way during a performance. To solve this annoying mystery, we suggest that you begin by checking out the wiring in the building. If the venue's lights and sound are run off the same electrical circuit, it can cause ground problems resulting in hum, also known as dimmer buzz. The same holds true for stage power and lights, or even stage power and house power.

Your first step in pinpointing dimmer buzz is to operate the lighting system, making it go from complete darkness to full brightness. Generally, if this affects the noise level, the intensity of the buzz will change, peaking between the one-quarter and three-quarters level of brightness (there may be no dimmer buzz at the lighting system's off or full position). If a definite change is apparent, your problem lies at the power source; find out where the lighting system is getting its power. If the lights and sound are feeding off the same breaker, try to move one of them to another circuit. If the simple test of operating the lighting system has had no significant effect on the buzz, your problem may lie with induction, the ability of cords to radiate energy to other cords within the proximity of several inches. One by one, unplug or switch out all the inputs to the board. If one in particular seems to make a change in the buzz level, it's likely that you've got a bad cord or the cord is touching (or is near) a cord from the lighting system. In that case, all you've got to do is separate the offending pair of cords. Tape them out of each other's way, if necessary. Be sure to check the cords onstage between mics, amps, and instruments: they are likely victims of induction.

The position of your amplifiers and speaker systems can also be the culprit. Danny Kapilian recalls the time he was mixing Steve Forbert at the Chestnut Cafe in Philadelphia; Steve simply could not get a persistent buzz out of his guitar sound. It turned out that his guitar amp was positioned right over the spot, where, under the stage, the power amps driving the PA system were situated. Steve merely had to move his amp over to lose the unwanted buzz.

If this problem is happening only with certain guitars, we suggest that you check out the various connecting links in the guitar system for loose wires or misconnected points. These would include the guitar cable, both of its ends (unscrew the housings and look inside), and the insertion jack in the base of the guitar.

Send your questions regarding club-style audio engineering to us at:
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sammy caine

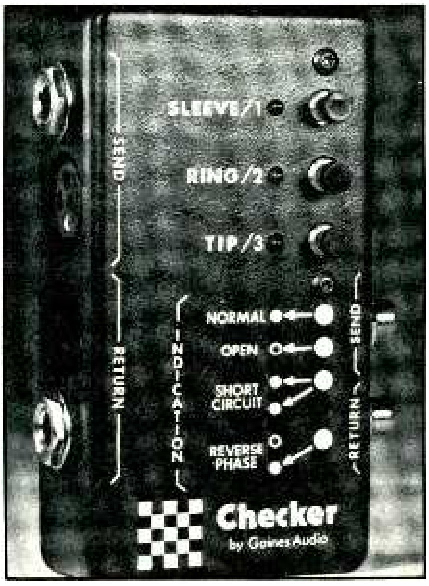
Product Profile

Kit Companies

This month's profile is a line of products made by several small companies. They are kits, and for small studios or home musicians/recordists, these kits offer a viable alternative to the high price of studio equipment. For a fraction of the cost of a ready-built piece of equipment, you can build it yourself. You end up with the satisfaction of knowing you did it yourself, and you may end up with a better product. Since you build the kit yourself, you also have a better understanding of the product, and if anything malfunctions, you can probably fix it yourself.

The quality of the product you get from a kit manufacturer can be superior to that of commercially available, completed products. They usually feature low noise and distortion, and a flat frequency response. The components included in these kits are of the highest quality and the design is straightforward and intelligent. Most of these companies design their kits to be easy to build—even if you are not very technically inclined.

There are a multitude of kit companies manufacturing kits for everything from color TVs and personal computers to keyboard synthesizers and outboard studio gear, but we will only look at those companies that manufacture kits relating to recording and musicians.



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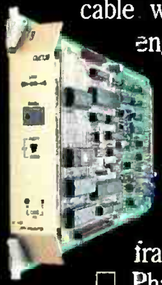

Otari's new EC-101 synchronizer module, when combined with the MTR-90 audio machine, creates an entirely new audio post-production system that uses a time-code-only link, via mic cable with the master. This unique "pre-engineered" combination offers performance well beyond that of any other audio tape recorder.

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So if your studio must stay current into the 1990's, or if your facility is now expanding into post-production, your timing is perfect. The breakthrough technology that gives you the best performing tape recorder in the world is here. And if you already own a MTR-90-II, an EC-101 is available as a plug-in option. From Otari: The Technology You Can Trust. Contact your nearest Otari dealer for a demonstration, or call Otari Corporation, 2 Davis Drive, Belmont, CA 94002 (415) 592-8311 Telex: 9103764890

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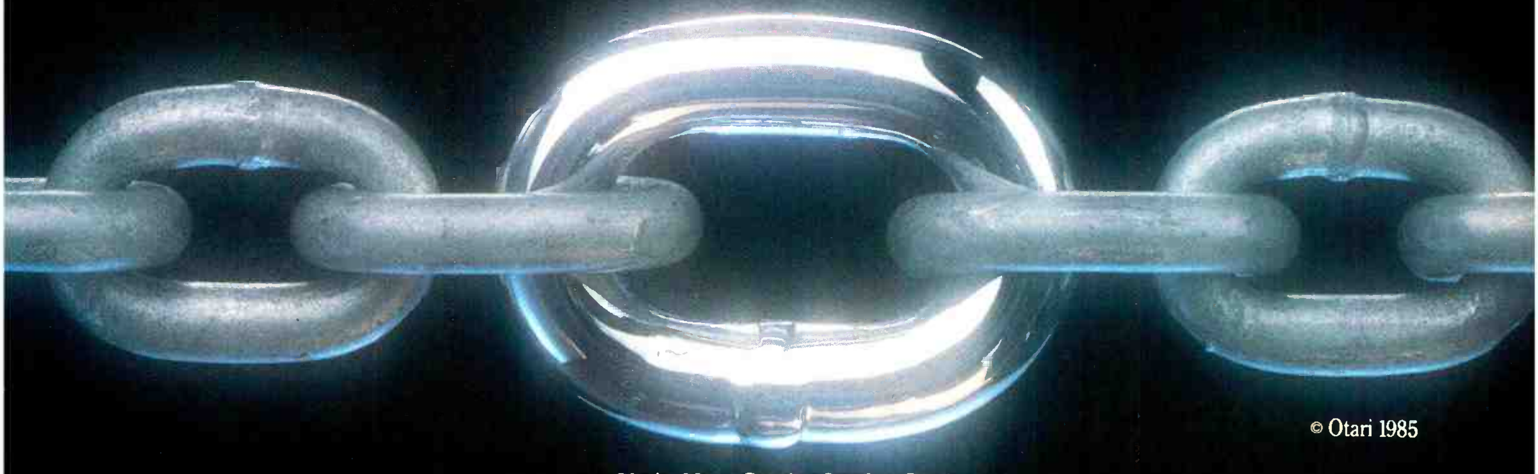

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Tri-Amp System C24-2000 DIRECT **$5469** List $9695
The new tri-amp C24-2000 Concert system produces a wall-of-sound that's earth shaking! Components include; the new 24 ch MX2422 pro stereo console, 2 dual 15" 2000M folded horn sub-woofers, 2 1220M mid-range enclosures, 2 R540H Carvin/Renkus-Heinz radial horns, 2 DCA800, 1 DCA300 power amps (1900 watts rms), and 1 XC1000 electronic cross-over. Also, our ultimate **SC24-2000 Super Concert** tri-amp system features twice the speaker components and sound coverage. DIRECT **$7699**

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Carvin Sound Systems. . .

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System S12-980 DIRECT **$1979** List $3895
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System S8-960 DIRECT **$1449** List $2695
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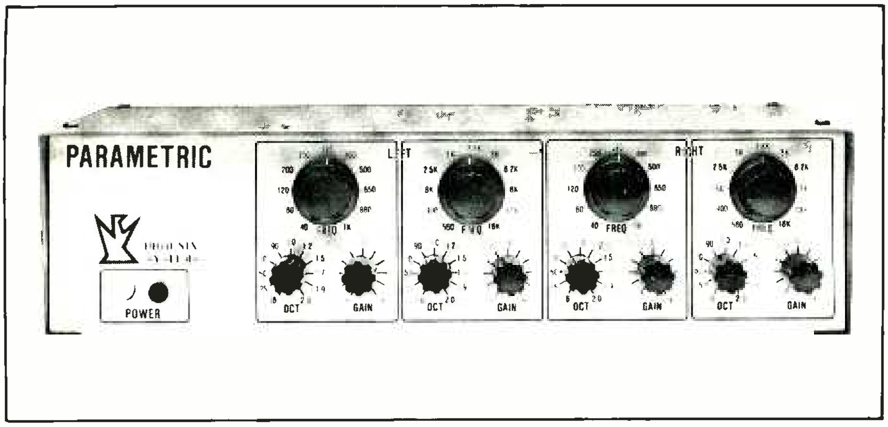

GAINES AUDIO

Gaines Audio offers kits to the pro-audio market for multitrack recording, sound reinforcement, and for the musician. Their kits are all based on pre-printed circuit boards and all components mount directly on the board so the kits are very simple to assemble. Even parts such as resistors and diodes are pre-cut and pre-formed to aid in construction.

The kit manuals are clear and concise and start with an introduction to the product and a complete parts list (with an accurate description of the part to help identify it). It gives an approximate assembly time range and a list of tools you need to complete the job. The manual also contains assembly drawings, stuffing guides and schematics, as well as easy to follow step by step construction details written in plain English. There is also a section on testing the unit with a list of 'most probable' causes in case of failure, as well as a complete section on use and operation of the unit.

Kits available from Gaines Audio include: compressor/limiter, noise gate, active direct box, dual power supply (suitable for use with any Gaines Audio product as well as other equipment), ten segment LED bar graph meter, active balanced input and output, and sine wave oscillator. All kits are also available completely assembled for a slightly higher cost.

PHOENIX SYSTEMS [Division of Soundware Corp.]

Phoenix offers a full line of kits to the high performance audio market. According to company president John Roberts, their kits are cost effective because of intelligent circuit design and a wise choice of components. Phoenix also offers a telephone help line for troubleshooting and repair advice.

The manuals contain background information on the product as well as hook up information. It also includes a section on use and applications that would be very helpful to someone without much background on the type of product. The assembly instructions include a short set of directions for soldering and some brief construction techniques. The actual step by step instructions are extremely easy to follow and if you check off each step as you do it, you can't get lost—even if you walk away from the project for a while. The instructions are divided into parts, which are then divided into subsections (1, 2, 3, etc.), which are then further subdivided into steps (a, b, c, etc.). A layout diagram and schematic is also included.

Kits available from Phoenix Systems include: parametric equalizer, noise reduction unit, regulated power supply and an audio test set. All kits are also available completely assembled for a slightly higher cost.

PAIA ELECTRONICS

As Paia's catalog indicates, they offer 'Electronics for the Creative Musician.' Paia sells kits for music and recording applications and they

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also sell all the kits included in Craig Anderton's book, "Electronic Projects For Musicians." They specialize in a complete line of modular and keyboard synthesizers, and manufacture a series of experimenter's kits, that are light on construction detail and heavy on theory and applications. Paia also offers a telephone technical service to assist in any project problems.

Paia's manuals are geared for the novice builder. It starts off with a few tips on soldering and a complete parts list along with color codes and drawings of the parts—so finding a particular part is easy. The construction operations are easily followed and illustrated with clear, straight forward photos and drawings. There are also sections on testing and calibrating, using the device and design analysis. Paia includes a removable set of assembly drawings as well for easy reference during construction.

Some of the kits available from Paia are: keyboard controllers and synthesizers (lead synths, guitar synths, drum synths, etc.), spring reverb, parametric EQ, hyperflange and chorus, limiter, synchronizer, stereo mixer, power supply, analog delay, linear digital to analog converter, noise gate, a full line of guitar effect pedals, guitar amps, tuner, and vocoder.

As I mentioned earlier, these are only some of the many companies manufacturing kits. There are many more and just about any product can be purchased in kit form. So if you feel technical enough, next time you're ready to make a purchase, consider the kit.

For more information on Gaines Audio, please circle number 50 on the reader service card.

For more information on Phoenix Systems, please circle number 51 on the reader service card.

For more information on Paia Electronics, please circle number 52 on the reader service card.

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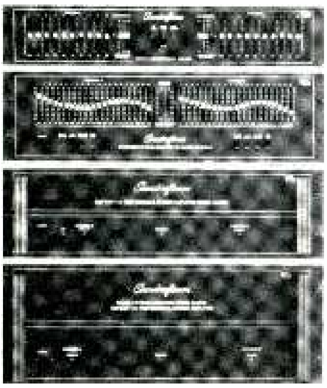

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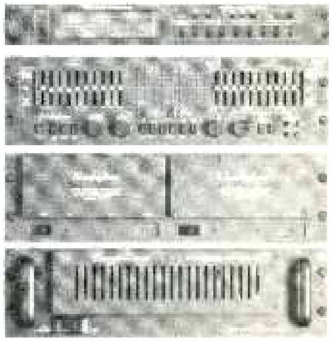

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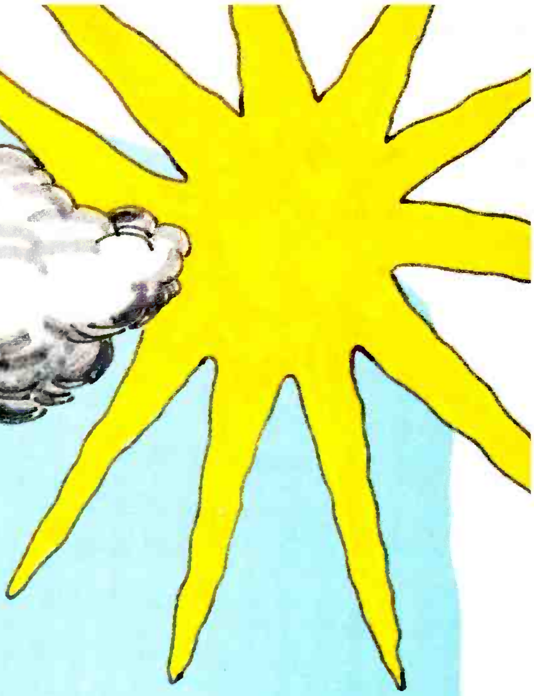

The Beach Boys

Riding a New Wave

America loves the Beach Boys, quite simply, because the Beach Boys *are* America—America personified, America suspended in time, America awaiting the perfect wave, America youthful, adventurous and romantic.

If Brian and Carl Wilson, Mike Love, Alan Jardine and Bruce Johnston had any doubts about who really owns the Beach Boys, the uncertainty was dispelled in 1983 when then-Interior Secretary James Watt, citing the "undesirable element" attracted to the group's shows, cancelled a Beach Boys performance on Independence Day in the nation's capital. The outrage was immediate and vehement; President Reagan bucked his cabinet member and backed the group. The Beach Boys were in, Watt on the way *out*. The Wilsons, Love, Jardine and Johnston *are* the Beach Boys, but, make no mistake, America *owns* the Beach Boys.

The Beach Boys, America's quintessential rock-and-roll band, America's longest-lived group, celebrate their twenty-fifth annniversary next year. Yet, every Beach Boys concert is a celebration, and the group has embarked on its longest and most eventful tour ever, having already performed a free show at the Washington Monument on July 4th and a few days later at Live Aid in Philadelphia; what's more,

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the greatest volume of call-in donations was generated during the Beach Boys segment.

The Beach Boys will forever be associated with Little Deuce Coupes, California girls and "Fun, Fun, Fun" in the sun. But behind all the innocence, youth and optimism of the group's classic songs of the '60s stood the childlike wonderment, musical genius and perfectionistic "Do It Again" (And Again, And Again) studio soundcraft of Brian Wilson. Using, at first, only three- or four-track facilities, Brian took the group's patchwork sound—Four Freshman-styled four- and five-part vocal harmonies and Carl's trebly Chuck Berry-derived guitar stylings—and embroidered around it his own intricate arrangements with the help of top-flight sessionmen like Hal Blaine, Jim Horn, Steve Douglas and Billy Strange. Grouping instruments according to texture and timbre into sections of strings or reeds or brass (not to mention the organ, theremin, cello, bells and chimes), Brian fashioned a "multicolored security blanket of sound" every bit as brilliant as the "wall-of-sound" productions of Phil Spector, his main influence and, ironically, competitor. And, ah, those voices! "Their vocal harmonies are unsurpassed," rocker Eric Carmen has said. "I think Brian was a French horn, Carl a flute, Al Jardine a trumpet, Dennis a trombone and Mike Love a baritone sax before their present incarnation as the Beach Boys."

But as times changed quickly in the '60s, as Camelot moved inexorably toward Altamont, so did the group that began in 1961 in Hawthorne, California, as the Pendletons and later Carl & The Passions. Pressure from Capitol Records for hit products, pressure from the Beatles and the British Invasion and pressure from the Wilsons' autocratic father, Murry, surely contributed to Brian's first emotional breakdown and his departure from the touring group in 1965. Working at home while the rest of the band, with new addition Bruce Johnston, took to the road, Brian began churning out new, more highly crafted and mature tunes, complex works whose lyric content left the marvelous eternal-adolescent messages of "Help Me Rhonda," "Surfer Girl," and "I Get Around" mired in the sand. In their place came such warm, personal musical missives as "God Only Knows" (with an evocative opening of French horn and quirky-sounding piano), "Wouldn't It Be Nice," "Sloop John B," "I Just Wasn't Made For These Times" and "Caroline, No."

Brian's craftsmanship and artistry jelled on 1966's *Pet Sounds*, rock 'n roll's first fully-realized concept album. Brian, the first member of a rock band to exert total control over his band's recordings, created a masterpiece with *Pet Sounds*, his high watermark as a producer-musician. That same year saw the release of "Good Vibrations," a multilayered paragon of pop and the group's only million-selling single. Expectations ran high that the follow-up LP, *Smile* (originally titled *Dumb Angel*), would capture Brian's rampaging creativity and do for the phonograph pop record what *The Jazz Singer* did for cinema. But *Smile* was never released (although snippets appeared here and there on the subsequent albums *Smiley Smile*, *Surf's Up* and *20/20*) and, instead, the release of the Beatles' *Sgt. Pepper's Lonely Hearts Club Band* captured the complete attention and adulation of the music world. *Sgt. Pepper* set new standards for musical architecture; *Smile*, sadly, stands as a sand-castle Sistine Chapel.

The Beach Boys continued to record (with several labels) over the next decade and a half, but the musical magic, like a message-in-a-bottle cast adrift at sea, lived more as a promise than as a potent reality. The myth coexisted with and sometimes exceeded the Beach Boys' musical output. Brian, by many accounts, became the Marcel Proust of rock, triumph having fermented into torment. Brian became, at times, either a recluse in his own room or a parody of himself, a composer stuck like his piano in a sandbox he helped to build. (Since 1976, however, Brian has surfaced on record and on tour; to date, though, he has shown only glimpses of his former brilliance.)

Though the Beach Boys' albums have been sporadic and creatively erratic in recent years—the group seeming to continue in spite rather than *because* of Brian—their live shows continue to be communal celebrations of their unique contribution to rock 'n roll. The group has survived Brian's emotional breakdowns, drinking bouts, Dennis Wilson's drowning death in 1983 and intra-group squabbling, but they continue to battle a double-bind. Carl told the *Detroit Free Press* recently: "We're damned if we do and damned if we don't. If we do a record that sounds like the others, it's the same old thing. And if we do something new, we'll be attacked because it doesn't sound like us."

The Beach Boys' new, eponymous album is their first in five years, their first with an outside producer in 24 years, and their first digital product. At once, *The Beach Boys* does (vocally) and does *not* (instrumentally) sound like the Beach Boys of old. Steve (Culture Club) Levine did the producing, and each Beach Boy contributes to the songwriting, including Brian, who co-wrote several numbers with his round-the-clock therapist, Dr. Eugene Landy. Boy George and Roy Hay chipped in with a tune, as did Stevie Wonder; drummer Ringo Starr guests on one selection. *Modern Recording & Music* caught up with Carl Wilson and Bruce Johnston in Chicago during their whirlwind tour. Their comments about the new album and the Beach Boys circa 1985 are revealing and, at times, diametrically opposite.

Modern Recording & Music: *The Beach Boys* is not only the band's first record in five years, but the first one with an outside producer and the first one recorded digitally. Three significant firsts.

Carl Wilson: It was an incredible process. This new technology blew our minds! It was a real revelation to us... It's bizarre to work with because you can sequence in the parts you're going to play—literally, the bass lines, the left hand, the right hand of the keyboard, note for note, bar by bar—so you're spending an hour or so just programming a section of the song. And you don't hear anything. Then you hook up the instrument you want to use and the sound you want. All of a sudden, it just comes together. I called it "scary monster" at first. It was just so different, instead of just having the cats play.

MR&M: Did making your two solo albums, Carl, prepare you in any way for doing *The Beach Boys*?

CW: Absolutely. Both of the records were made while I was working with the [Beach Boys] group. One was done in late '80 and the other was done during a touring break. I did take a year off from the group; I stopped working with the group in February 1981. During that time I was gigging with my own band, playing some clubs and also opened the show for the Doobie Brothers. That really made me a better team

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player [with the Beach Boys]. I really care. It occurred to me at that time the guys [in the Beach Boys] just weren't into producing a real high energy level, didn't care too much and weren't able...So I had to get busy and make some music.

MR&M: I've long considered you the conscience of the Beach Boys, besides, of course, the dominant instrumentalist [guitar] in the band.

CW: I just couldn't bear it. The last ten days of that summer tour of 1980 were really strange. It just went to sleep. There had been this incredible charge and thrill and rush attached to every time we performed for twenty years, and then all of a sudden to have it go almost into a coma was really disturbing to me.

MR&M: And you had no inkling of this previously?

CW: No, not prior to that.

MR&M: After being away from the Beach Boys for a year, what prompted you to come back? Did you return of your own volition or did the circumstances that precipitated your departure improve?

CW: Everybody had agreed, by that time, to record and rehearse. I wanted to put into the work the energy it deserved. The thing we found is that the more you rehearse a tune, the more present and fresh it becomes. It doesn't get older, it gets *newer*, when you work at it. When you are doing excellence with a tune that could be as ancient as the hills, it brings the tune present into this moment, it doesn't make it older.

MR&M: Did the digital recording process on the album suggest—or necessitate—any live-performance innovations on this concert tour?

CW: Oh, yeah. We're doing synthesized bass now, which is really a pleasure for us to hear a different bass sound onstage. Ed Carter, our bassist, really loves playing the keyboard bass.

MR&M: As the lead guitar player onstage, does the synthesized bass change your conception at all? I'm sure its time-keeping is perfect.

CW: It's a little bit different. On some of the synthesizers onstage, the touch is such that you almost have to come down a little early to be on time. Whatever you do, you don't want to play your notes early...We have a synthesizer because of the bass, which is a Roland, I believe.

MR&M: I understand from Steve Levine, the producer, that your tunes on the album, which were essentially guitar-based as written, were translated onto keyboards. How did that process work?

CW: I sat with Steve and Julian Lindsay, his associate, and I would show them the parts, which he would transpose on the keyboards. Sometimes he would write the notes out, and sometimes the verbal communication was so quick that the delivery went right into his head and through his fingertips into the sequencing. Sometimes I would play it for him and we would make notes and chord charts...The tunes I contributed to the album were all written on guitar.

MR&M: Did you find that the transition to keyboards altered the essential character of the tunes in any way?

CW: No. The overtones and harmonics you get with the guitar are different with the keyboard. With the synthesized instruments you're unlimited in terms of what you can add or subtract from any given note. For instance, on the song "Where I Belong," that's a guitar voicing through the Kurzweil...it's sequenced in. They're exactly the right notes, but it's a different sound. We're really open to whatever presents itself in sonic terms. We took what we got and kind of liked it.

MR&M: You also play some keyboards yourself on the album.

CW: On "Where I Belong" I used the Planet of Isis on the DX1 display. It's a sound I heard, and I said, "We've got to use this on the record somewhere!" It's the real high thing in between the phrases in the chorus.

MR&M: Had you played the DX1 previously?

CW: No, the first time I saw it was in their studio in London. It blew me away! For hours I was just playing all these sounds on the DX1.

MR&M: When you wrote your tunes on guitar, did you imagine how they'd turn out ultimately on the synthesizers?

CW: I imagined them real different. I heard it as an acoustic sound, then opening up into whatever it was. I like to keep that open until we record; that way, I don't get locked in on something and limit it. That bass sound on "Where I Belong"—I thought, "Oh, boy. This is going to be really heavy-handed." It sounds like a sawtooth thing for a minute, and the sound keeps changing with the filters. Yet, it worked okay.

MR&M: Several recording studios were employed to make this album—Red Bus and CBS in London, and Westlake Audio in California. Which parts of the album were recorded where?

CW: The basic tracks were started in London in June. During June and July I think we did nine basic tracks there. I was there for a week in June, and Brian went over in July for a couple weeks. Then Steve went to CBS in London to do the drums for "Maybe I Don't Know." Perhaps he did [the drums by] Ringo [Starr] at CBS London for "California Calling." The tracks for "Getcha Back" and "She Believes in Love Again" were done at Westlake in LA. Then we did all the vocals at Westlake. And Bruce did the verse section of "She Believes in Love Again" back in London, I believe. Steve mixed the album down at Red Bus and brought it back to L.A., where we did the remix at Westlake.

> **One of the strengths of the album is that we got these really clear, clean and crisp sounds on the track, and even large sounds, and the vocals are real warm.**

MR&M: What capabilities attracted you to each of these studios?

CW: Westlake was designed by Westlake Audio; Eastlake Audio, over in Europe. Hidley designed both rooms. They're almost identical, though Westlake has a larger control room. They've very similar in electronics. The board at Red Bus is an MCI, and the board at Westlake is a Harrison. The monitor systems were similar. Of course, we didn't use the big monitor system at Westlake; it was just too *honky* (whistles).

MR&M: What about the capacity for vocals at Westlake?

CW: It seemed to be an easy enough room to work in. I'll tell you who's got a great room for vocals—Studio 1 at Cherokee in L.A. You can hear yourself, you can hear the pitch...

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When you record digitally, it's a different approach to engineering. Gordon [Milne] played a real big part in the recording of our voices. One of the strengths of the album is that we got these really clear, clean and crisp sounds on the track, and even large sounds, and the vocals are real warm. Some people would accuse the digital process of making records sound cold or sterile.

MR&M: The vocals are indeed welcoming and endearing.

CW: The vocal sounds we got have a warmth. I felt it was a nice marriage of a classic approach to singing harmonies—we've basically done the same thing since 1962—and...the fabulous technology that's available.

MR&M: The Beach Boys, it seems to me, took a risk by accepting a new recording process after you'd been away from the recording scene for five years.

CW: ...We didn't really spend that much time on this record. I wish when we did a record we had six full, full months to work on it. This album cost about 400G's. That's a lot of money considering that we spent a day or possibly two on each track... But the cost isn't outrageous.

MR&M: The snare-drum sample done by Levine at the racquetball club on one tune reminds me of Brian's recording techniques in the '60s, when he'd record sounds in an empty swimming pool, for instance.

CW: Most of the drum sounds [on the album] were computer-generated and sampled through the Fairlight.

(The new album) is very consistent with our history of trying lots of new things. I'm just saying that, in hindsight, I don't think our tracks have the power of some of our older tracks.

Over at the Century West Club, they went over to the spa one day and Steve got this idea to record a snare drum in one of their racquetball courts. That echoey snare-drum sound on "Getcha Back" is a sample taken out at the racquetball court. [In the '60s with Brian] we used to do background parts at all sorts of places. Steve has a real playful quality, very similar to Brian, naturally.

MR&M: How is Brian adapting to all the new technology?

CW: He was a little bit indifferent at first. It kind of freaked him out a little bit, but not seriously. It's a real wig to walk into a room and see all this stuff. And you program the music *note by note*, and you don't really hear it for a while. It's technology, you know?

MR&M: In retrospect, it's amazing how well Brian organized sounds into groups of instruments, wrote charts and verbalized his needs to the assembled musicians. While his "pocket symphonies" in the '60s have been duly acclaimed, it seems his interpersonal skills have not been sufficiently noted.

CW: What you want to keep in mind about some of Brian's more evolved works, such as *Pet Sounds*, is that there were no sound effects involved in that record; that was simply arranging. There were records being made at the time that were starting to play a lot of sound effects and stereo stuff. Chuck Britz did a very good job recording the [*Pet Sounds*] stuff. The truth is that Brian did not have the skill in terms of notating the music, so he would communicate verbally to the musicians.

MR&M: Brian's ability to group instruments into sections during the '60s was ground-breaking. I wonder if the new technology would enable him, or anyone else for that matter, to replicate or update that process.

CW: What he would need to do in that case, and what I trust he is entering into, is *learning* the technology. After he went through that initial couple days of (pause)...I don't know if *indifference* is the most accurate word to use...he was really *for* it. He made a conscious choice and said, "Hey, I want to jump into this thing head first and really have a good time making these records."

MR&M: Besides yourself and Brian, how did the other band members take to the new recording process?

CW: I think everyone was really pleased. We all had a real healthy respect for it. It's a little more time-consuming [than analog recording]. By far, the longest stretch of the album was done recording the vocals. There was a little resistance at first. It took everybody a couple of days to get into the motion of it; they were working with a new person. It took a lot of care to do.

MR&M: I'm told that Steve Levine had some difficulty recording the ensemble vocals on this record, and as such, he recorded you and Bruce first, then the others separately, so he could get the right harmony tunings. Is that so?

CW: What is so about that is that we were *not* getting great performances by everybody. It was difficult to get a good performance, pitch-wise and blend-wise. What you do is make a "slave tape." On two channels you have a stereo mix of the track; it's the earphone mix. We would get locked into an earphone mix where we could not really get the particular mix we each might like to have to hear the very best we can hear. Michael [Love], for instance, has to hear his voice very loud. Some of us like more drums; some of us like different instruments for pitch. We weren't getting a good earphone mix at first. I have a very good sense of pitch, and I seem to be able to nail things real fast, so what we did was I would record my part, then we would stack. Once my part was on, people could hear the pitch. It was really a pitch thing. The difference was truly night and day.

MR&M: Carl, you sound so sanguine and open-minded about all these incredible innovations in recording. Looking back to the '60s, the Beach Boys were for a time the most innovative studio rock group in the world. By today's standards, on what is today very primitive equipment, you made everlasting music. So I'm surprised that you say there was no resistance and arrogance by the group, where some might be expected.

CW: I think there's a feeling of gratitude that pervades the group that people have been so patient in allowing us to go through *all* the stuff

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we've gone through to come back full circle to a beginning again with this record... I had a ball with those guys.

MR&M: Over the past twelve to fifteen years, the Beach Boys have been far more prolific as a touring band than as a recording band. Whereas in recording an album, it's not necessary to have everyone in the band in the studio at all times, onstage it is, of course, necessary. Has this duality caused any confusion within the band?

CW: Perhaps. *I'm* very clear on it. Personally, I think everybody should be involved with *everything*, all the time. Of course, not *every* moment. Everybody's view is valuable; everybody brings something different in terms of viewpoint into the project.

MR&M: Generally speaking, and with the Beach Boys in particular, when is a song overproduced?

CW: That's a really good question.

MR&M: Are there greater dangers of overproducing with the new digital technology today?

CW: No, there's less danger [today]. You can get more on tape now. You can put more information on a thing with less overload. But we've suffered from [overproducing] from time to time. I guess it becomes too much when it gets in the way of the song.

MR&M: Then again, at the top of this conversation, you mentioned that your guitar-written tunes were translated to keyboards. From a layman's viewpoint, it seems like the new digital equipment might offer a greater possibility for overproducing.

CW: See, with the new technology, the sounds carry usually less harmonics. You put twenty acoustic instruments on a recording, and you put twenty synthesized sounds, and the acoustic instruments will have far more overtones and harmonics. So that puts more energy on the tape. I think you would tend to overload, first, with acoustic instruments. If you can make it with three, four or five instruments, I think that's really great.

MR&M: Sure, because it allows room for the listener to add his own input.

CW: Exactly. There's a lot of air in it that way. You can get a great sound on four things a lot easier than you can get a great sound on fifty things.

MR&M: That's one of the great things about the Beach Boys' old songs. Driving in the car, you have the choice of singing along with five different voice parts. Or, you can harmonize something of your own.

CW: There's really something about that simplicity that works wonderfully.

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Bruce Johnston, fresh-faced and darkly handsome, was a staff producer at Columbia Records in 1965 when he received the fateful call to replace Glen Campbell, who had replaced Brian Wilson on tour. Brian, the Beach Boys' mastermind, had decided to quit the road and stay at home to write full-time. As a bass player, songwriter, producer and falsetto-singer à la Brian, Johnston filled the bill perfectly. And what started as a temporary alliance turned into a 20-year, off-and-on (mostly on) membership in the Beach Boys. Now 41, Johnston produced the group's last album, 1979's *L.A. (Light Album)*, but vows that he "volunteered *never* to produce another Beach Boys album." Johnston, who says he has "outgrown the record business but not the Beach Boy business," did, however, turn the group onto Culture Club producer Steve Levine, whom he credits with making "Smokey Robinson-meets-the-Beach Boys albums in the '80s." Levine had engineered the fourth album by the group Sailor, which Johnston produced in 1977. "I'm going to do my most candid interview of the year," Johnston, who was interviewed alone, after Carl Wilson, told *Modern Recording & Music*. Straight off, he stated that his comments about the new Beach Boys album would in certain respects veer "180 degrees" from Carl Wilson's. He then immediately set sail against the prevailing wind.

Modern Recording & Music: Besides being the Beach Boys first album in five years and first with an outside producer, *The Beach Boys* is also the group's first digital recording. How did the band face that challenge?

Bruce Johnston: I don't think very well. I don't think we pulled together as a band on this album. We, again, have little solo singles. I'm not very happy with this album. I think we've let everybody down. I think the singing sounds *wonderful*, a couple of songs are terrific, and on a technological level it's absolutely *perfect*, but none of it is *right*. And we really don't deserve to have a hit album with this album.

MR&M: Did this happen because the group had been away from the recording scene so long?

BJ: No, because we don't get in the same room as a band and talk about music. We all show up at the studio with separate thoughts and we pursue them, then we put our voices on top.

MR&M: But that [individual approach to composing] isn't a totally new development for the band.

BJ: No. It's carrying our problems forward in the '80s, digitally.

MR&M: Steve Levine told an editor of *Modern Recording & Music* that he had difficulty producing this record because, with the exception of you and Carl, the other members were (A) fearful of the new technology, and/or (B) would rather be on the road. Did the other members have to be convinced—or shamed—into being in the recording studio as a band?

BJ: No. I don't think anyone was fearful of the technology at all. On the Fairlight level, when you sample the bass, a lot of the track sounds don't have the power of, say, a Huey Lewis track. I would like our [basic] tracks to have a little wider dynamic range; I think there's a very sterile glossiness in our tracking.

MR&M: What accounts for this sterility?

BJ: I think possibly the electronics in the sampling of some of these wonderful instruments.

MR&M: Because it isn't a group, *playing* record?

BJ: There are no instruments—we really didn't play on the record. But it doesn't matter that we didn't play. The reproduction, sonically, of a lot of the sampled instruments doesn't have the dynamics that you have when you actually play them. On the other hand, I like the idea of digital recording, because I know how well we sing. I have no complaint with our singing and the sound of it; that's always been our strong point. But I do have a problem listening to the tracks up against what I hear on the radio. And they just don't have the *power*. Our vocals have more power than the tracks; it's very mismatched.

MR&M: Steve Levine noted in an interview with this magazine that he had some trouble recording the ensemble vocals on this record. Is that so? And if so, how?

BJ: I think it might have been more of a lack of patience on the part of the band and the producer. When you've had really great four- and five-part singing for twenty years, you just don't run into the studio and expect [immediate results]; you have to warm up a little to it. Steve really

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didn't wait sometimes for that to happen.

MR&M: I'm told that it took a while to get the vocal levels and harmony tunings down.

BJ: It wasn't any different than it was before, yet we were flexible enough to try it other ways to please the producer. I'm not unhappy with Steve at all. I'm just unhappy with our creative contribution as a band. Frankly, none of us are as good as Brian Wilson. And it doesn't seem as though Brian runs the race as well as he used to. When you get down to it, the guy who should be in command is Brian Wilson, and he's not.

MR&M: The question is: Is Brian capable of taking command at this point? Or willing?

BJ: If you compare him to an athlete who has played at Wimbledon, he can still play great tennis, but I don't think he'd make it to the semi-finals. It's the continuing problem that Brian has with his music. Don't think I'm negative. I'm not blaming anybody for the reason I'm unhappy about our album—I'm blaming *all of us*. We just can't seem to get in a room to be a *band*. Yet we have these fantastic [live] shows, fantastic tours, a great deal of money twenty-five years down the road almost, a great deal of income from this wonderful legacy Brian gave us that's the cornerstone of what we are. It's also our problem because of radio—it's very difficult for people to play a new Beach Boy record when they know they can pull it off by playing two minutes and twenty seconds of "Surfing U.S.A."

MR&M: Is it not possible that some aspect of the Beach Boys powerful live show can be brought to bear in the recording studio, short of a conventional live album?

BJ: It's absolutely possible and it's got to be as dynamic on all levels. We are capable, as performers, of singing as well if not better than we did twenty years ago; it's obvious on the album. On the level of songwriting, production, performance and then the follow-up by a label, we are totally capable of doing something as important as "What's Love Got to Do With It." But I don't think that piece of material has surfaced on this album.

MR&M: Was there much aforethought among the band members as a whole about this record?

BJ: No. I think we all thought about it independently. I think people are so *relieved* that there's a Beach Boy album where we sound really great and that there are things that resemble songs. I think we're getting reviews just because people are glad we're still around.

MR&M: There is a certain amount of gratitude and nostalgia value. And the Beach Boys cachet, the singing, is excellent.

BJ: I think people are grateful that Don Ameche, who is seventy-seven years old, is still able to drive off a diving board and star in a movie [*Cocoon*]. We're getting a little of that. But I think we're better than that. I think that just because we can still stand and breathe doesn't mean that people should applaud.

MR&M: But you must realize that the Beach Boys don't own the Beach Boys—the American public does. In the same way, the Coca Cola Bottling Co. does not own Coca Cola—the American public does, as witnessed by the furor over the planned recall of *old* Coke.

BJ: Absolutely. And if Coca Cola tried to sell Dom Perignon or invent it, the public wouldn't let them. So that's our problem. Being a songwriter, I'm *painfully* always aware of this. The *only* answer is to write or find the greatest piece of material since "What's Love Got to Do With It." That kind of song just makes you make that great record. And that's what I hope we run into by the next album.

MR&M: On the new album, Bruce, you play the Kurzweil 250. Do you also use it as a composing tool?

BJ: No, I think the Kurzweil piano sound is not even close to [what I might have achieved] had I been able to use the Yamaha in the studio. In terms of making tracks, there are lots of other sounds on the Kurzweil. There was an acoustic bass and guitar sounds, but for my ear, I think the nine-foot Yamaha grand in the studio is a helluva lot better sounding instrument than the piano that is claimed to be a concert grand on the Kurzweil chip.

MR&M: Do you compose at home on a Yamaha grand?

BJ: No, I've got a Yamaha C7 at home. I'll compose on any kind of keyboard. I compose on a 149 dollar Casio. When you want an acoustic-piano sound, you might as well use an acoustic piano. You might as well have it beautifully mic'ed in stereo going to the digital machine, rather than reproducing the sound on the Kurzweil in the studio. Now, if you're home or on the road, there might be some use for the Kurzweil. What I'm trying to say is that you should try and decide, on a technological level with the instruments, what sound is really going to sound better by the time it hits the digital tape-recording process. Are you going to have an inferior sound but feel like you are making more progress because you use these digital chips that are in these instruments or even the synthesized stuff in the DX7? Or do you make the choice to go for the best acoustical piano sound, for instance? In other words, you don't want to let the technology have control of *you*. You have to wade through it and see what sounds best. Certain kinds of recordings really shine with the [new] technology.

MR&M: Do you think the technology took control of *you* on this recording?

BJ: At times. I don't think the tracks have much power. One track, "Maybe I Don't Know," sounds pretty good. You don't want to use technology for technology sake.

MR&M: You want the technology to enhance the tunes, which must take primacy.

BJ: Absolutely. First of all, you have to decide: What is the right song and what feeling do I want to use from it? You don't want to turn your back on all the developments; you want to *use* them. But you have to know when they're not right for you.

MR&M: Having made this new record with the digital process, you have plenty of time before the next album to become even more familiar with the options and possibilities of the equipment.

BJ: I don't think they are that necessary for what we need to do. We really need to get closer in our uptempo songs to the tracks you hear on a lot of the Huey Lewis records. I think we're a noisier band [than the record indicates]. All the track sounds on this album are too polite.

MR&M: Which groups in your estimation, Bruce, are doing innovative things in the studio, all the while maintaining the much-needed grit you speak of?

BJ: I have to mention Huey Lewis again, because that says it all for me in tracking and the way I hear Beach Boy sounds, vocally. I can just *hear* the grittiness of Carl's lead voice with Huey Lewis's tracks and our [group] vocals. Sonically, there's a wider range, as opposed to a deeper range, on a subsonic range that you can't get

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with these instruments we're talking about.

MR&M: The lack of conventional instruments on this album is a departure from the old days of the Beach Boys when Brian masterminded incredible sounds from sections of musicians.

BJ: It's not different in terms of being experimental. [The new album] is very consistent with our history of trying lots of new things. I'm just saying that, in hindsight, I don't think our tracks have the power of some of our older tracks.

MR&M: What advantages did the Fairlight afford the band on this LP?

BJ: On "Crack at Your Love" it's really cute. There are some interesting drum sounds we put in it that we recorded in a [racquetball] court; we used that on "Getcha Back."

MR&M: Are you referring to the snare-drum sounds?

BJ: Yes. I think it's really great to be used to pick up auxiliary sounds. But for my taste, I would use a lot of the technology on an auxiliary level ...For sounds you need to make for radio, for the Beach Boys, you need to take along your vocal sound, your level of songwriting, but you also don't want to do a parody of your past. You want to show some growth without losing the reason people liked you in the first place. So I'm only speaking about the Beach Boys.

MR&M: Has the experience of recording this record manifested itself, and in any way improved, the already-potent force of the Beach Boys live performance?

BJ: I don't know (in a disgusted tone of voice).

MR&M: Carl mentioned, for instance, that you're now using some bass synthesizer.

BJ: Yeah, but we have so many [old] songs we're expected to do, and we can't do a whole show of new songs. Using the keys, rather than the bass, for a couple songs gives it a different sound. It really makes it close to the [sound of] the record, for instance, on "It's Getting Late" and "She Believes in Love Again." But, again, it's just auxiliary and alternative things that we're doing [onstage]. Our keyboard synthesizer section has a lot of stuff; it's kind of tucked away so it's not there for its own sake. It brings back French horns and strings on "God Only Knows." [Synthesist] Mike Meros has a lot going. He reproduces a lot of things we did on the album. So we're totally up to date on one corner of the stage.

MR&M: Using the racquetball club to record the snare sounds seems to harken back to Brian's exploratory openness in the '60s, when he recorded sounds in an empty swimming pool, among other places.

BJ: It's kind of Brian-in-box, in terms of the Fairlight. It's just another way of doing something Brian did 20 years ago.

MR&M: Does Brian see that?

BJ: Absolutely. He loves all this stuff.

MR&M: Bruce, let's turn a moment to the three recording studios used to make *The Beach Boys*—two in England and one in California. As a producer yourself, and as a member of the Beach Boys, how well did these facilities serve the needs of the record?

BJ: Unfortunately, one of the studios was in Los Angeles—a very unhealthy place to record, unless you're out by the beach. The studios really aren't very remarkable; they're state-of-the-art. The only remarkable thing was the reproduction of what we did using the Sony 24-track digital machine. Most of the recording studios in the world are really behind the digital technology; the mixing boards are a little old-fashioned. The next thing is that you'll have boards that are totally at the same level as the digital machines. The studio of the future will have a very large control room and a very small studio; it will be the reverse of the studios you see today.

MR&M: That lends itself, potentially, more to a producer's medium than to a player's medium.

BJ: Exactly. That was a problem with our album. One of the biggest problems we had is that we never saw our producer; we only heard him. There was ninety percent no eye contact because he was behind a lot of equipment. On a human level, we could never see his reaction.

MR&M: That state of affairs runs contrary to the human dynamic behind the great Beach Boys albums of the '60s, when Brian was in the studio coaching the musicians and sections, feeding off musicians, taking suggestions. I don't mean to sound anachronistic, but from a listener's standpoint, I think that face-to-face interplay helped make those albums sound so lively.

BJ: Absolutely. And that was totally missing on this record. And we as a band were very ununited in terms of our songs. Look, we have an album that has sold several hundred thousand copies, [the vocals] sound great, and I'm telling you I'm unhappy with it. I'm telling you, truthfully, that we have to use this technology to *our* advantage; we can't let the technology use us to *its* advantage.

MR&M: You've stated, Bruce, that you've volunteered *not* to produce any more Beach Boys albums. But you're so forthright and vehement in your criticisms of the album and the band, as a whole, that it seems incumbent on you to pull on the other members' coattails.

BJ: I just like to go to bed at eleven o'clock. I have three little boys at home, and I refuse to sit in Hollywood making records late at night.

You finally knew that all these artists who talked about changing the world did something. I think I could have retired from the business after Live Aid.

MR&M: At what point, Bruce, is a song overproduced?

BJ: Your ears will tell you when it sounds too full, when there are too many things to listen to. When you add orchestras on top of [basic] tracks with lots of vocals, it tends to get a little messy.

MR&M: And, I infer from your comments earlier that you do not consider the new album overproduced.

BJ: No, this album is very sparse. Our band, as a band, did not get together and put into it what we should have, even though we all did sing. And the tracks lack a lot of punch. This album does not hold up against what I'm hearing on the radio for punch.

MR&M: And it wasn't because the

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band members were frightened about the new technology?

BJ: I don't think anybody was frightened, I think they were disappointed. We wound up hearing less than what we heard before, when we were on a roll, in terms of power in the tracks. You can turn anything up loud and have power, but there's the kind of power that's there even when you hear your tracks down low.

MR&M: There is a visceral quality irrespective of volume.

BJ: Absolutely. When you get down to it, Brian Wilson is missing, even though he's here. The Brian Wilson of "California Girls" and "Fun, Fun, Fun"—Brian Wilson the musician who happened to be called a producer—that genius is missing. He set our standards so high that it's hard for us, even with an outside producer at this point, to live up to what he created.

MR&M: I've never met or spoken with Brian, but isn't the fabulous work he did in the '60s the product of the times, of growing up, of pressure from Capitol Records for a hit product and from his overbearing father?

BJ: I'm sure that's part of it. Bruce Springsteen is probably never going to be any bigger than he is right now. Sometimes when you've stumbled into something like that, there's a certain kind of energy that just takes you and pulls you and shakes all these wonderful things out of you in a very short time...I know, vocally, we can always deliver like Tina [Turner] did if we have the right situation; I don't know where it's going to come from. On a technological level, I view everything that's out there as something we can consider to add on to whatever we're doing...You have to use your ears and find out what's best for you. A lot of times when you use the keyed instruments that make sounds, you might be taking a shortcut with your record that in the end will have no heartbeat.

MR&M: Is the answer to the Beach Boys' recording problem, then, to be found in the band itself or necessarily in an outside producer?

BJ: I think all of us are really talented singers and writers, but I don't think any one of us has the quality, which Brian exercised, to lead the band into the future. Without Brian, or a Brian, we run the risk of becoming a musical, movable theme part. However, this is the record business, and there are those people out there who do have the magic but don't have the artistry available to them to articulate musically the magic they have. So here we are (pause), unless Brian decides totally to be Brian Wilson again.

MR&M: I return to the cruel, concurrent irony that the Beach Boys are so electric and alive in concert.

BJ: We *are*, but in the studio we are not equal to what we are on the road. Yet, with just pure sounds we make vocally, we are kind of freeze-dried until someone pours boiling water on us.

MR&M: Some might say, "Well, being a fabulous live band is enough."

BJ: But it *isn't* enough for any of us. We're not complacent about it. I didn't feel much magic making this album, but I know on a technological level, it's a beautiful-sounding album. People really do like it, but I know we can be better than that. You see, I come from the school that says, "Don't record until you really are sure that the songs are right for the project, and where are the hit songs? And don't put one of yours in if there's another one that's better." I know what's more important than any kind of technological development—*great music*. Otherwise, this stuff is meaningless.

MR&M: When groups say they're going into the studio, the laymen usually thinks they are going in to lay down songs they've worked out, polished and rehearsed. In actuality, however, often the creative process doesn't begin *until* the band enters the studio.

BJ: I just don't think that people can afford to start getting creative at 200 dollars an hour. I think you'd better have a helluva good blueprint before you even get to the studio; otherwise, if you get lucky enough to go double platinum, you may still be in the red...It keeps a lot of engineers and people who make Kurzweils and Fairlights in business. The 200-dollar-an-hour clock ticks away while you're experimenting in the studio with these keyboards. A lot of people should really go to school first and learn how to use all this technology on an instrument level before they even go in the studio; otherwise, their budget is going to go crazy.

MR&M: Could the answer be that the group must spend more time together as a group outside the recording studio yet in a creative setting?

BJ: Al Jardine wants us to go away for two or three weeks up to his ranch, just do music in one of his barns, just interact and write. We're friends but we're real disjointed. You're talking about a big business. I think we'll do that before the next album. The album just isn't up to our standards, even though it sounds great. It's not anybody's fault on the technological level—the problem is with the human being...I expect something [on record] at least equal to our best nights on the road.

MR&M: Speaking of live shows, what was your impression of performing at Live Aid?

BJ: That's probably the easiest show I ever did in my life. It was probably the only *real* day in the industry since I've been in it. It was financed beautifully. As you know, no one received any money for it. Everything was perfect. You finally knew that all those artists who talked about changing the world *did* something. I think I could have retired from the business after Live Aid. I finally saw everything we thought about come to life and work.

MR&M: It sounds like you know what the Beach Boys must do and *not* do in order to get back on track as a recording entity.

BJ: I want to be as good as we are. I don't want to simulate it and have people love us just because we're walking around and breathing and able to sing some of those old songs... This album is a warm-up. In terms of the next record, this was a demo to get us pointed in the direction of making great records and really being interested in it.

MR&M: Would you like the next album to be more of a *playing* record?

BJ: I want to play with the *band* on the next record. I want to make basic tracks with the band, and *then* plug the technology in. I want to put the *breathing* back in the record. The tracks are too dull...You don't feel the rhythms speed up just a hair as you change keys. It's not that I'm faulting Steve. I think that people expect a little bit more of a human sound inside the Beach Boys as far as the tracks go. I want to play with a *band*; I want to *watch* the drummer. I don't necessarily want the drum sounds limited to acoustic drums; I'd like them to trigger a Simmons or some of the other stuff available. I want to play with sidepeople in the studio. I don't want to miss the magic. And I don't *ever* (laughs) intend to produce another Beach Boys' record. I want to protect our history, but I want to add onto it. I don't want to coast.

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MODERN RECORDING & MUSIC

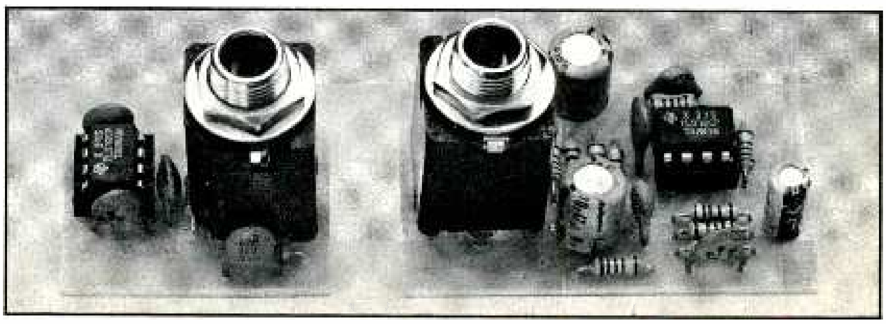
Tenth Anniversary Issue

jon gaines

Building Active Balanced Input and Output Circuits

This helpful project is reprinted from the January 1985 issue.

Buying the equipment is easy compared to plugging it in. A glance around the average studio control room reveals a mind-boggling assortment of input and output connectors; nominal line levels, impedances, and balancing schemes. Interfacing products from various manufacturers can be a minor nightmare for studio owners and engineers, especially if your



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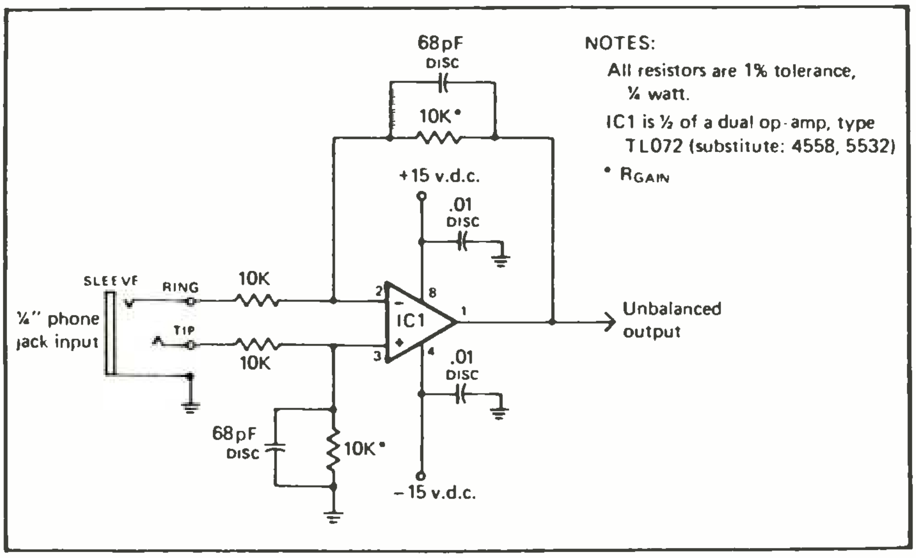

Figure 1. Balanced Input.

talents lie in areas other than electronic design.

Possibly the most frequently asked question in pro audio is: "How do I balance an unbalanced input (or output)" that is followed immediately by: "How can I match the levels of these two pieces of equipment?" For example, you want to plug the unbalanced, -10 dBm output of your 4-track recorder into a mixing console that is designed for balanced +4 dBV inputs. Perhaps the situation is complicated by the fact that the recorder has a high impedance output which you want to route around the control room through some 40 feet of wire before connecting it to the console, a combination that can result in significant deterioration of high frequency response.

The most common solution to this type of problem is to grab as many adapters and plug-in matching transformers as you can find and hope for the best. Unfortunately, the transformer as line balancer/unbalancer, level matcher/impedance matcher is often less than ideal, since these various objectives usually collide with each other. In the case of the unbalanced 4-track recorder, a transformer will do a good job of balancing its output and "stepping down" its output impedance, but it will simultaneously drop the recorder's output level by roughly 20 dB, which is exactly the opposite of the gain change desired. If you then try to make up the gain by turning up the mixing console, noise performance will be degraded. There is a better way. This article describes two common audio circuits: the differential input amplifier and a balanced output/line driver, both of which are in use by major audio manufacturers and may even be a part of some of the equipment you already own. They are the electronic equivalent of matching transformers, with the added advantages of easy gain setting and very low cost. The circuits are easy to build, easy to apply, and may be attempted by anyone with even modest electronic experience. Between the two of them, you can solve virtually any interference problem.

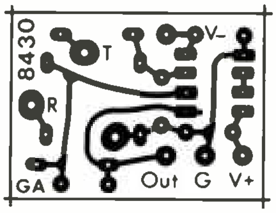

Figure 2. Actual size printed circuit board artwork for active balanced input circuit. This view is from the copper side of the circuit board. The pads marked T, R, and S are provided for use with a ¼" phone jack, or may be connected to an XLR connector, pins 3, 2, and 1 respectively.

Balanced Input

The balanced input amplifier shown in *Figure 1* is drawn as a unity gain amplifier (input level-output level) which accepts a balanced input signal on a ¼-inch tip-ring-sleeve (stereo) phone jack and converts it to a low impedance, unbalanced output which can then be connected to the input of an unbalanced piece of audio equipment (the phone jack could just as easily be an XLR type connector, of course). By simply changing the value of the two resistors marked R_{Gain}, the circuit can be made to have a loss or a gain (lower volume or higher volume at the output). Typically, this circuit will be used when you have a line level balanced source feeding an unbalanced unit that

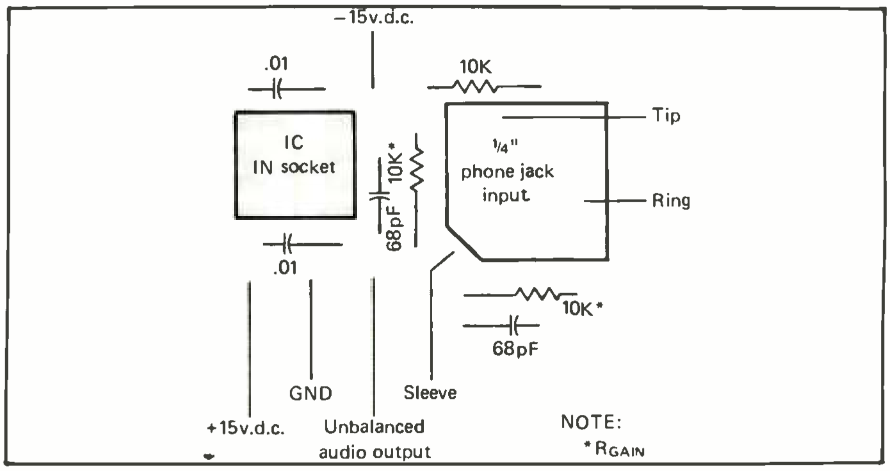

Figure 3. Component stuffing guide for active balanced input. This view is from the component side of the circuit board.

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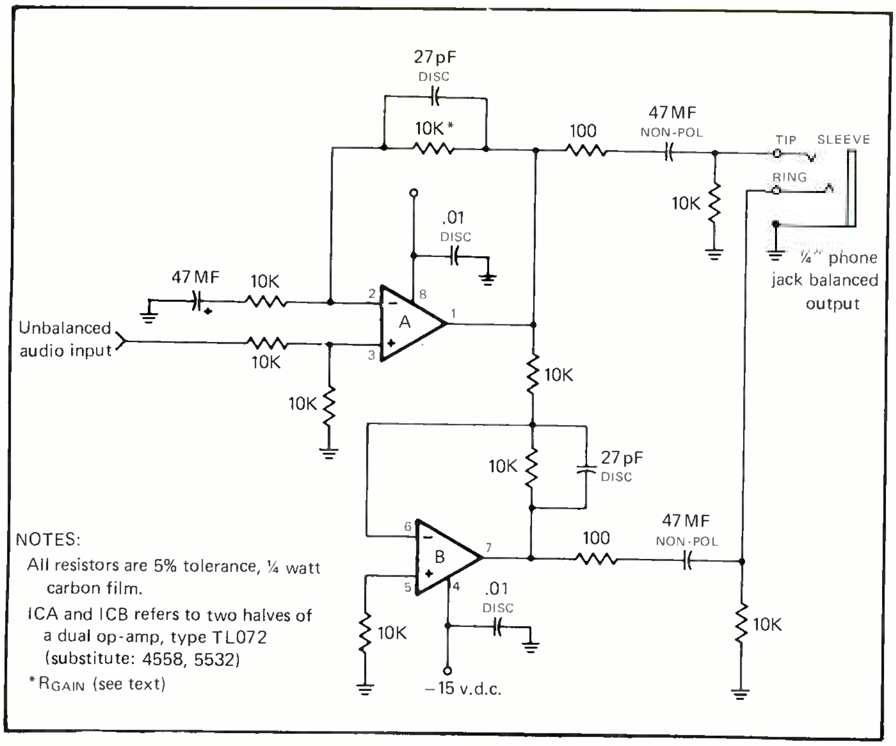

Figure 4. Balanced output circuit.

wants either the same or a lower level.

For example, if you want to connect the +4 dBm balanced output of a mixing console to the -10 dBV input of a tape recorder, you must fill two requirements: you need to unbalance the console's output and get rid of 14 dB of level. So, to make this circuit work for this application, simply consult *Table 1*, which indicates that the value of the two R_{Gain} resistors should be 2K ohms, substituted for the 10K resistors shown in the *Figure 1* drawing.

R_{Gain}	Level Loss
10K	0 DB (Unity)
6.8K	- 4 dB
4.3K	- 8 dB
3K	-10 dB
2K	-14 dB
1.3K	-18 dB
1K	-20 dB

Table 1. Commonly available resistor values used to set gain in the Balanced Input circuit of *Figure 1* (± ½ dB).

The circuit has an inherently high input impedance (20K ohm) and a low output impedance (around 10 ohms), making it an ideal impedance matcher for all modern audio equipment. The days of rigid impedance matching are gone, thankfully, and this high-Z input/low-Z output structure is now fairly universal.

Suppose that you want to use the gain and impedance matching features of this circuit, but you don't need the balanced input feature, i.e., you're feeding it from an unbalanced source. Just take the leg of the circuit which is shown connected to the ring of the input jack and connect it to ground instead. The gain of the circuit is unchanged. This ability to switch back and forth from balanced to unbalanced input operation adds to the versatility of this circuit. When used in conjunction with a tip-ring-sleeve phone jack as shown in *Figure 1*, the input will be balanced if fed from a balanced tip-ring-sleeve phone plug, and will automatically unbalance itself if an unbalanced tip-sleeve (mono) phone plug is inserted. No re-wiring needed; the connector does the work.

Other characteristics of this circuit are: low noise, low distortion, perfect frequency response, and phase integrity; it does not reverse the phase of the input signal.

Note that the circuit is shown using all 1 percent tolerance resistors. This is suggested to maximize the common mode rejection ratio of the input amplifier. Put simply, this is an expression of the circuit's ability to reject noise that gets into your cables through the atmosphere.

Examples of this are radio station signals, television sync buzz, magnetic hum, light dimmer noise, and, lately, emissions from personal computers. The closer the matching of the resistors in this circuit, the better the amplifier's ability to reject such noise. Note that there are two pairs of resistors here: the two 10 K input resistors form one pair, and the two R_{Gain} resistors form the second pair. It is not important for the resistors to be within 1 percent of their marked value; the point is to match the two resistors within the pair so that they are within 1 percent of *each other*.

Before you rush off to track down precision components, let me offer an easier (and cheaper) approach. If you have access to a digital volt-ohm meter, you can select your own 1 percent resistors out of a bag of ordinary 5 percent parts. Typically, more than half of the resistors in a batch of 5 percent types will turn out to be within 1 percent of their marked value. And even if you have to settle for unsorted 5 percent parts, you'll still end up with better than 20 dB of common mode rejection, more than adequate for most line level applications.

The two 68 pF disc capacitors improve stability and filter out high frequency noise. This value is not critical, and you can use anything from 50 to 100 pF here, depending somewhat on the type of op-amp chosen.

Finally, the .01 MF disc caps shown attached to the power supply lines are a standard protection against noise and oscillations. Locate them close to the IC.

The balanced output circuit of *Figure 4* uses two op-amps, one non-inverting and the other inverting, to

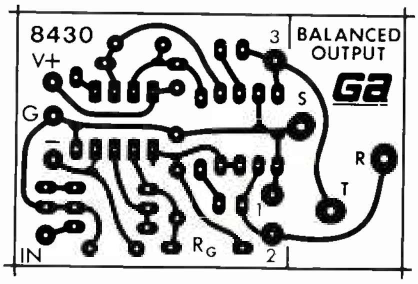

Figure 5. Balanced Output printed circuit board artwork, actual size. This view is from the copper side of the board. The large oval pads at the right are for use with a PC mounted phone jack. If not needed, this section of the circuit board may be cut off, reducing the overall size of the card. The pads marked 1, 2, and 3 would then be used as the outputs, shield, minus, and plus, respectively.

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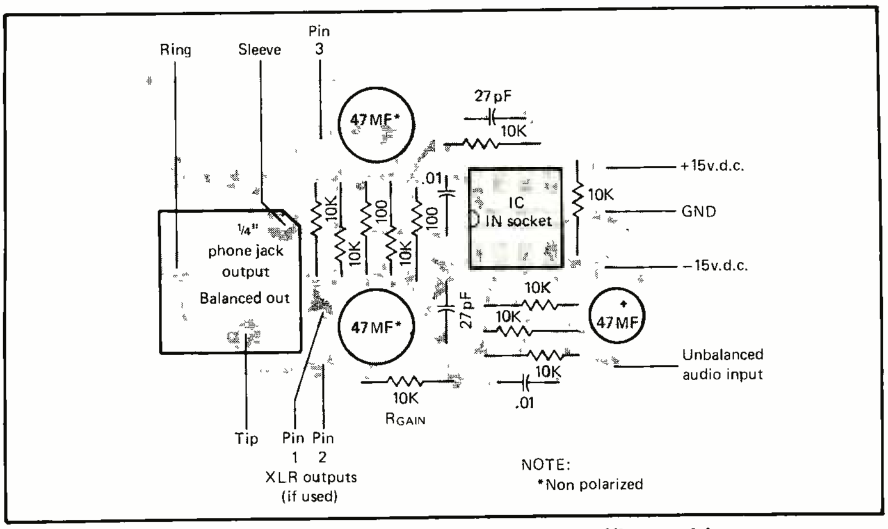

Figure 6. Balanced output component stuffing guide. This view is from the component side.

R_{Gain}	Balanced Output Level	Unbalanced Output Level
0 ohms	+ 0 dB*	- 6 dB
6.8K ohms	+ 4 dB	- 2 dB
15K ohms	+ 8 dB*	+ 2 dB
24K ohms	+10 dB*	+ 4 dB
39K ohms	+14 dB*	+ 8 dB
56K ohms	+16 dB*	+10 dB
100K ohms	+20 dB	+14 dB
200K ohms	+26 dB*	+20 dB

*unity

Table 2. Resistor values for balanced output circuit gain setting.

produce a symmetrically opposed pair of output signals which can then drive a balanced line and, at the other end of that line, feed a balanced input. As shown, it has a gain of 6 dB as measured across the outputs of both amplifiers, and the gain of this circuit can be modified by changing a single resistor, again called R_{Gain}.

Typically, this circuit will be used at line level and will either have unity gain or some increase in gain. As an example, suppose that you want to convert the unbalanced -10 dBV output of a tape recorder to a balanced 0 dBm output. Just consult *Table 2*, which specifies a 24K ohm resistor for R_{Gain}.

Like the balanced input circuit discussed earlier, the circuit of *Figure 2* allows you to choose balanced or unbalanced operation, depending on your need. To use it as an unbalanced level matching circuit, simply leave the bottom amplifier disconnected from the ring of the output connector and use only the first stage of the circuit. You can still take advantage of the op-amp's buffering and impedance matching characteristics, and you can still set the gain to any level you like. Consult the third column in *Table 2*, "Unbalanced Output Level," to determine the correct value for R_{Gain}. Note that the level seen across two amplifiers is always 6 dB hotter than the level across one amplifier, thus the need for two level columns in *Table 2*.

Since both the tip and ring connectors of the output jack are "hot" in this circuit, you must never plug in a mono phone plug; the barrel of the plug would short the second op-amp to ground, which could eventually cause a failure. The two 100 ohm resistors in series with the outputs will protect the op-amps against momentary shorts, and also improve the .tability of the circuit when driving long cables and other capacitive loads. Cable runs and snakes of up to 200 feet are perfectly reasonable loads for this circuit. The 47 MF non-polarized capacitors on the outputs are intended as DC blocking caps. They guarantee that no matter what might go wrong ahead of this point, no damaging DC voltages will ever get any farther.

None of the component values in this circuit are particularly fussy, so normal 5 percent tolerance resistors will be adequate, and cap values can be adjusted slightly if necessary.

Power

These circuits require a standard ±15V DC power supply, either regulated or unregulated. If you own any contemporary audio equipment, you probably already have several regulated power supplies inside your gear. If you are adding these balancing circuits to existing equipment, you may be able to build them right into the product and share its power supply. Since the circuits draw a miniscule amount of current, you're not likely to overload the power supply. I should caution you, however, that some manufacturers may not honor their warranty when such a modification has been made, so it would be prudent to consult them before doing any work on your in-warranty units. In any case, you can certainly build these circuits as an outboard interface box with an independent power supply. Shielding is not especially critical, but a metal chassis would still be a good idea.

SPECIFICATIONS

BALANCED INPUT
Noise: -110 dBV
Frequency Response: DC to 100 kHz, ±0.5 dB
Maximum Input/Output Level: +20 dBV
THD: less than 0.01%
CMRR: typically 70 dB
Current Required: 8 mA max, at full output into 600 ohms

BALANCED OUTPUT
Noise: -105 dBV
Frequency Response: 5 Hz to 100 kHz, ±0.5 dB
Maximum Input Level: +20 dBV
Maximum Output Level: +26 dBV
THD: less than 0.01%
Current Required: 16 mA max. at full output, 600 ohm load

PARTS AVAILABILITY
Parts for this project are available from Gaines Audio, PO Box 17888, Rochester, New York, 14617, as follows:

Complete parts kit for the balanced input circuit as drawn in Figure 1, $7.95. Printed circuit board only, $4.65.

Complete parts kit for the balanced output circuit as drawn in Figure 4, $9.35. Printed circuit board only, 5.25.

Prices are postage paid. NY residents please add tax. Visa, Mastercharge, money orders accepted.

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This helpful guide is reprinted from the July 1984 issue.

bruce bartlett

dB or not dB

Confusion runs rampant when it comes to decibels. How is a dB different from a dBV or a VU? How do you match component levels? How do you compare microphone sensitivities? In this article we'll try to clarify these subjects. The reading involves a little math, but it is essential if you want to be knowledgeable about this aspect of audio. Even experienced professionals may find some surprises here.

Basics

First, let's define the term "level" as used in a recording studio. Originally, "level" meant power, while "amplitude" referred to voltage. Today, the term "level" is used to denote voltage or sound pressure, although the terminology is not entirely correct. Still, it is important to understand both definitions in order to communicate.

Audio level is measured in decibels (dB). The original, classic definition of decibel was: 10 times the logarithm of the ratio of two power levels, or:

$$dB = 10 \log(P1/P2)$$

These days it's common to use "dB" to refer to voltage ratios as well. Hence, the following formula:

$$dB = 20 \log(V1/V2)$$

One dB is the smallest *change* in level that the human ear can hear—the just-noticeable difference. Actually, the just-noticeable difference varies from 0.1 dB to about 5 dB, depending on bandwidth, frequency, program material, and the individual. But 1 dB is generally accepted as the smallest change in level that most people can hear.

Sound pressure level, signal level, and change in signal level are all measured in dB. Let's look at each of these.

Sound Pressure Level

Sound pressure level is the pressure of sound vibration, measured at a point. It's usually measured in dB SPL (decibels of sound pressure level).

The higher the sound pressure level, the greater the perceived loudness (see *Figure 1*). The quietest sound we can hear, the threshold of hearing, is 0 dB SPL. Average conversation at one foot is to 70 dB SPL. The average

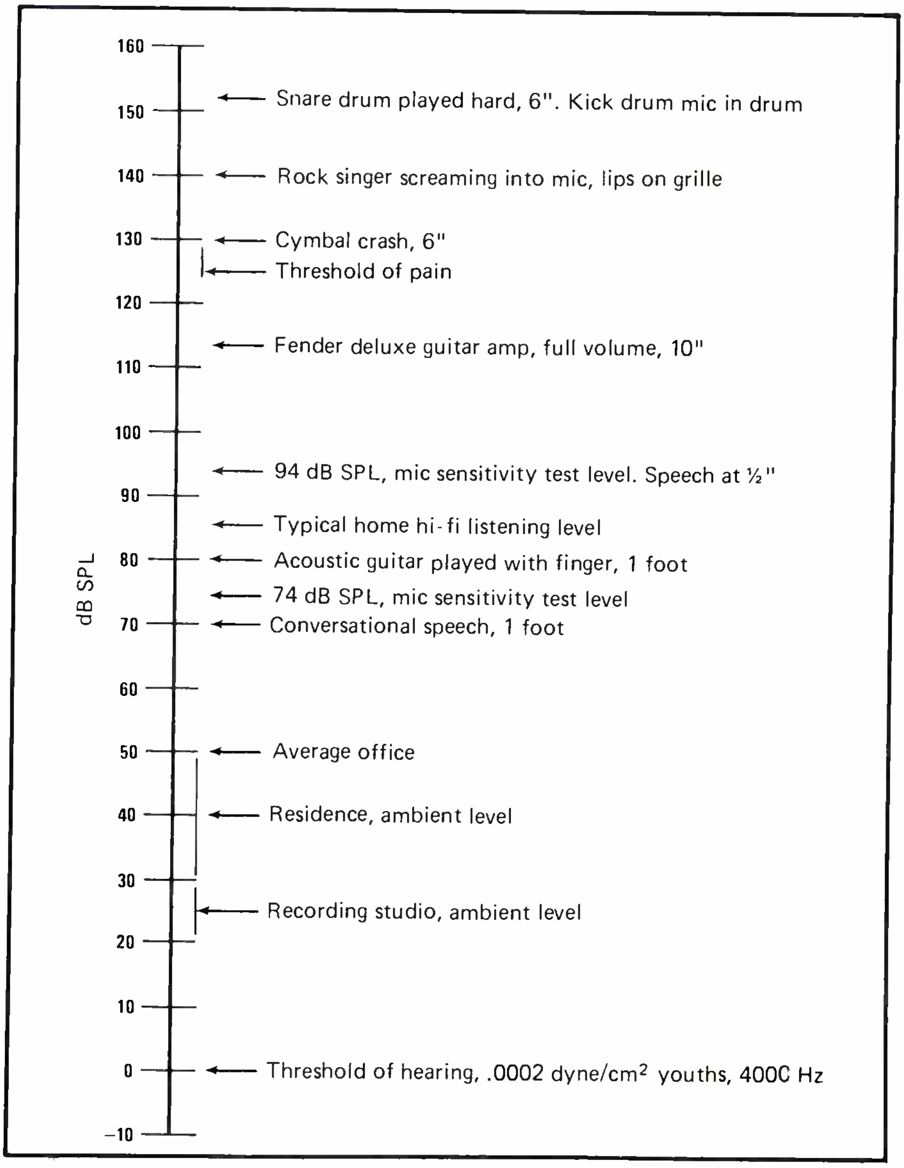

Figure 1. Chart of Sound Pressure Levels.

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home-stereo listening level is around 85 dB SPL. The threshold of pain—so loud that the ears hurt—is 125 to 130 dB SPL. A 10 dB increase in SPL is considered by most listeners to be twice as loud.

Sound pressure level, in decibels, is 20 times the logarithm of the ratio of two sound pressures:

$$\text{dB SPL} = 20 \log P/P_{ref.}$$

where:

P = the measured sound pressure in dynes/cm² and

$P_{ref.}$ = a reference sound pressure: .0002 dyne/cm² (the threshold of hearing).

Signal level

Signal level is also measured in dB. Signal level, in decibels, can be expressed in various ways:

dBm or dBl: decibels referenced to 1 milliwatt.

dBu or dBv: decibels referenced to .775 volt.

dBV: decibels referenced to 1 volt.

Let's explain each one of these.

If measuring signal power, the decibel unit to use is dBm.

$$\text{dBm} = 10 \log P/P_{ref.}$$

where:

P = the measured power and

$P_{ref.}$ = the reference power, 1 milliwatt.

For example, let's convert .01 watt to dBm:

$$\text{dBm} = 10 \log P/P_{ref.} = 10 \log .01/.001 = +10$$

So, .01 watt is 10 dBm (10 decibels above 1 milliwatt).

Now let's convert .001 watt to dBm:

$$\text{dBm} = 10 \log P/P_{ref.} = .001/.001 = 0.$$

So, 0 dBm = 1 milliwatt.

Any voltage across any resistance that results in 1 millwatt is 0 dBm.

$$0 \text{ dBm} = V^2/R = 1 \text{ milliwatt}$$

where:

V = the voltage in volts and

R = the circuit resistance in ohms.

For example, .775 volt across 600 ohms is 0 dBm. One volt across 1000 ohms is 0 dBm.

Some voltmeters are calibrated in dBm. The meter reading in dBm is accurate only when measuring across 600 ohms. For an accurate dBm measurement, measure the voltage and circuit resistance, then calculate:

$$\text{dBm} = 10 \log \frac{V^2/R}{.001}$$

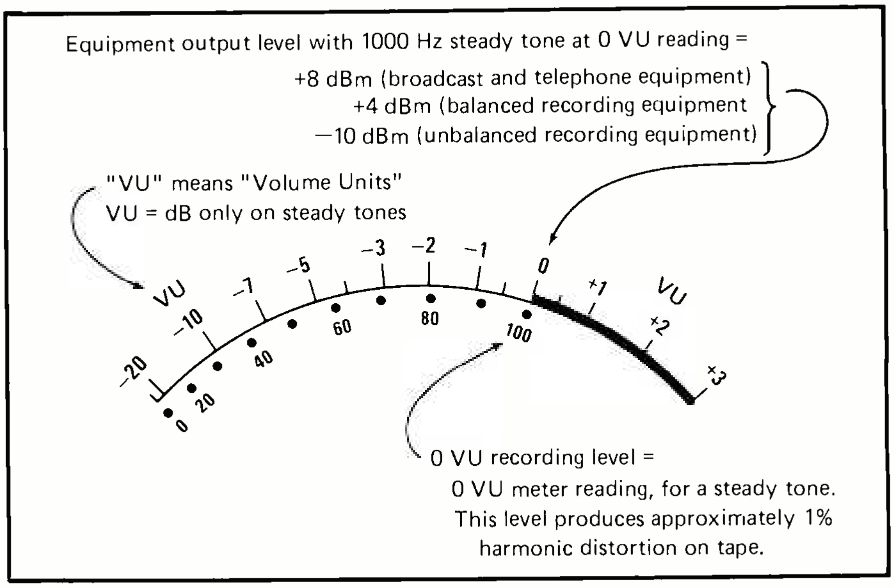

Figure 2. VU meter scale.

Another unit of measurement is called dBv or dbu. This means decibels referenced to .775 volt. The ".775 volt" figure comes from 0 dBm. 0 dBm = .775 volt across 600 ohms, where 600 ohms used to be a standard impedance for audio connections.

$$\text{dBv} = \text{dBu} = 20 \log V/V_{ref.}$$

where:

$V_{ref.}$ = .775 volt.

Signal level is also measured in dBV, or decibels referenced to 1 volt.

$$\text{dBV} = 20 \log V/V_{ref.}$$

where:

$V_{ref.}$ = 1 volt.

For example, let's convert 1 millivolt to dBv:

$$\text{dBV} = 20 \log V/V_{ref.} = 20 \log .001/1 = -60$$

So 1 millivolt = -60 dBV (60 decibels below 1 volt).

Now let's convert 1 volt to dBV:

$$\text{dBV} = 20 \log 1/1 = 0$$

So 1 volt = 0 dBV.

To convert dBV to voltage, use the formula

$$\text{volts} = 10^{\frac{\text{dBV}}{20}}$$

Change in signal level

Decibels are also used to measure the *change* in power or voltage across a fixed resistor. The formula is:

$$\text{dB} = 10 \log P1/P2$$

or

$$\text{dB} = 20 \log V1/V2$$

where P1 is the new power level and P2 is the old power level; V1 is the new voltage level and V2 is the old voltage level.

For example, if the voltage across a resistor was .01 volt, and it changed to 1 volt, the change in dB is

$$\text{dB} = 20 \log V_{new}/V_{old} = 20 \log 1/.01 = 40.$$

So the change in voltage is 40 dB.

Doubling the power results in a 3 dB increase; doubling the voltage results in a 6 dB increase.

Summary:

$$\text{dB SPL} = 10 \log P/P_{ref.}$$

where

$P_{ref.}$ = .0002 dyne/cm.²

$$\text{dBm} = 10 \log P/P_{ref.}$$

where

$P_{ref.}$ = 1 milliwatt.

$$\text{dBV} = 20 \log V/V_{ref.}$$

where

$V_{ref.}$ =1 volt.

$$\text{dBu} = 20 \log V/V_{ref.}$$

where

$V_{ref.}$ = .775 volt.

$$\text{dB} = 10 \log P1/P2 = 20 \log V1/V2.$$

The VU Meter and 0 VU

A VU meter (originally called VI meter) is a voltmeter of specified transient response, calibrated in VU or "volume units." It shows approximately the relative volume or loudness of various signals.

The VU-meter scale is divided into volume units, which *are not necessarily* the same as dB. Volume Units = dB *only* with steady-state sine wave tones. That is, 1 VU = 1 dB only when a steady tone is applied. When transients or complex waveforms are measured on a VU meter, the marks

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on the meter scale do not correspond to dB.

A zero meter reading is usually called "0 VU," although this definition is not completely accurate. "0 VU" refers to a level, not a meter reading. Still, you have to know both definition to communicate.

By "0 VU," most recording engineers mean a 0 reading on the VU meter. When the meter on your mixer or recorder reads "0" on a steady tone, your equipment is producing a certain level at its output. Different types of equipment produce different nominal levels when the meter reads "0," as shown below and in *Figure 2:*

Broadcast and telephone equipment:
 0 VU reading = +8 dBm.
Recording equipment (balanced):
 0 VU reading = +4 dBm.
Recording equipment (unbalanced):
 0 VU reading = -10 dBV.

When a tape operator says to a mixing engineer, "Send me a 0 VU tone," he or she means, "Send me a tone that reads 0 on your VU meter." The actual level isn't too important, because the engineer receiving the tone just wants to calibrate his or her equipment (by setting the tone to a 0 VU meter reading).

A little-known fact is that 0 dBm = 0 VU. A *level* of 0 VU is not the same as a *meter reading* of 0 VU. When the meter reads 0 VU, the actual level being produced at the output of balanced recording equipment is +4 dBm or +4 VU. Confused? Don't worry about it.

A 0 VU *recording level* (0 on the record level meter) is the normal operating level of a recorder that produces the desired recorded flux on tape. A "0 VU recording level" does not mean a "0 VU signal level."

With a VU meter, 0 VU corresponds to the recording level that produces about 1 percent distortion on tape, at a frequency from 333 to 1000 Hz. Distortion may be slightly above or below 1 percent, depending on the tape used.

There's another kind of recording-level meter used mostly in Europe: the peak program meter (PPM). It responds very rapidly to peak program levels, making it a more-accurate indicator of true recording levels. The PPM is calibrated in dB, rather than VU. Unlike the VU-meter reading, the PPM reading does not correlate with perceived volume.

A VU meter responds too slowly to track rapid transients accurately, so it usually reads lower than the actual peak level. For example, if you're recording drums at 0 VU on the meter, the actual peak level may be 8 to 14 dB higher. So, whenever you record instruments having a high peak-to-average ratio, such as drums, piano, percussion, or horns, record at -8 to -14 VU to avoid saturating the tape. Some mixers and recorders include an LED that flashes on peak overloads, which helps you set recording levels more effectively.

Balanced vs. Unbalanced Equipment Levels

Generally, audio equipment with balanced (3-pin) connectors works at a higher nominal line level than equipment with unbalanced (phono) connectors. There's nothing *inherent* in balanced or unbalanced connections that makes them operate at different levels; they're just *standardized* at different levels.

Shown below are the nomi
mal) input and output levels
two types of equipment:

Balanced: +4 dBm (1.23
 or +4 dBu)
Unbalanced: -10 dBV (.316 volt)

When a balanced-output recorder reads 0 VU on its meter with a steady tone, it is producing 1.23 volts at its output connector. This voltage is called +4 dBu when referenced to .775 volt, or +4 dBu when referenced to 1 milliwatt. When an unbalanced-output recorder reads 0 VU on its meter with a steady tone, it is producing .316 volt at its output connector. This voltage is called -10 dBV when referenced to 1 volt.

Interfacing Balanced and Unbalanced Equipment

There's an 11.8 dB difference between +4 dBm and -10 dBV. How did we get that? By referencing both voltages to 1 volt:

+4 dBm = 20 log 1.23/1 = +1.8 dBV.
-10 dBV = 20 log .316/1 = -10 dBV.

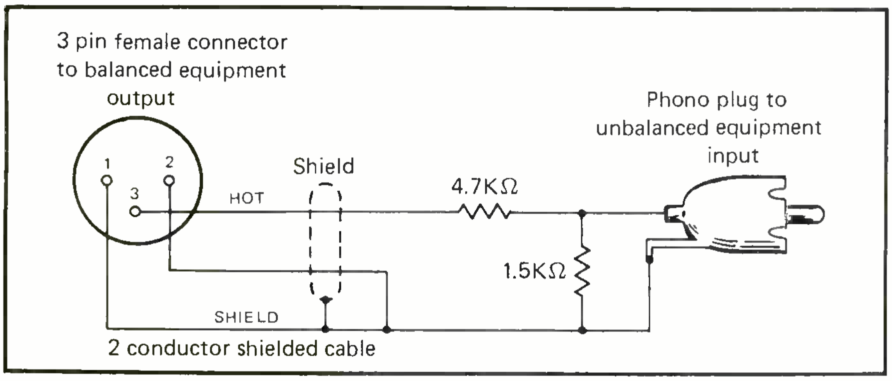

Figure 3. 12 dB pad, which matches output level of balanced equipment to required input level of unbalanced equipment.

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So, +4 dBm is 11.8 dB higher in voltage than -10 dBV.

Connecting a +4 dBm output to a -10 dBV input will most probably cause distortion, because the signal peaks of the +4 equipment may exceed the headroom of the -10 equipment. To attenuate the level 12 dB when connecting the two types of audio gear, you can use a pad as shown in *Figure 3*. It converts from balanced to unbalanced and reduces the level 12 dB. (You may have to substitute a stereo phone plug for the 3-pin connector.)

However, you don't always need that pad. Many pieces of equipment have a "+4/-10" level switch. You just set the switch to the nominal level of the connected equipment.

Microphone Sensitivity

Here's another confusing area concerning decibels: microphone sensitivity. A microphone-sensitivity spec tells how much output (in volts) a microphone produces for a certain input (in SPL). A high-sensitivity microphone puts out a stronger signal (higher voltage) than a low-sensitivity microphone, when both are exposed to the same sound pressure level.

Microphone sensitivity is especially confusing because it is specified in many ways:

1. dBV per pascal
2. dBV per 10 dynes/cm^2
3. dBV per microbar
4. dBV per dyne/cm^2
5. dBm per 10 microbars
6. dBm per 10 dynes/cm^2
7. dBm, EIA

We'll explain each of these. First note that:

10 dynes/cm^2 = 10 microbars = 1 pascal = 94 dB SPL

1 dyne/cm^2 = 1 microbar = 74 dB SPL

A typical microphone sensitivity specification might be: Open circuit voltage: -75 dB re 1 volt per microbar. That means, the mic produces -75 dBV, unloaded, when exposed to a sound pressure level of 1 microbar (74 dB SPL). You put 74 dB SPL in; you get -75 dBV out.

Another way to express the same sensitivity is: Open-circuit voltage: -55 dBV/pascal. That is, the mic produces -55 dBV, unloaded, when exposed to a sound pressure level of 1 pascal (94 dB SPL). You put 94 dB SPL in; you get -55 dBV out.

Here's still another way to specify the same sensitivity: Power sensitivity: -55 dBm per 10 microbars. In other words, the mic produces -55 dBm into a matched load, with an SPL of 10 microbars (94 dB-SPL). "Matched load" means that the load impedance equals the microphone impedance. If the mic impedance is 250 ohms, the load impedance of the mic preamp input is also 250 ohms. This is unlikely to occur in practice; usually the load impedance is at least 7 to 10 times the mic impedance.

"EIA Sensitivity" is expressed in G_M. It's seldom seen, but is useful for calculating the mic's output in dBm for a given input in SPL.

SPL + G_M = dBm output into a matched load.

Here are some typical microphone sensitivities:

Condenser mic: -65 dBV/microbar
Dynamic mic: -75 dBV/microbar

Expressed another way:

Condenser mic: -65 dBV per dyne/cm^2
Dynamic mic: -75 dBV per dyne/cm^2

Expressed another way:

Condenser mic: -45 dBV/pascal
Dynamic mic: -55 dBV/pascal

Expressed another way:

Condenser mic: -45 dBm/10 microbar
Dynamic mic: -55 dBm/10 microbar

You can't directly compare the sensitivities of two microphones specified in different ways. You have to convert them to the same reference, using these simple conversion formulas:

dBV/pascal = dBV/10 microbars = dBV/microbar + 20 dB.

dBm/10 microbar = dBm/10 dynes/cm^2 = dBV/microbar + 20 dB (if mic impedance = 250 ohms).

dBm/10 microbar = dBm/10 dynes/cm^2 = dBV/microbar + 22.2 dB (if mic impedance = 150 ohms.)

G_M (EIA) in dBm = dBV/microbar - 71.76 dB (assuming the mic's rated impedance is 75 to 300 ohms).

Let's use the formulas above to compare the sensitivities of two mics rated in different ways.

Mic #1: -70 dBV per microbar (impedance = 250 ohms).
Mic #2: -55 dBm per 10 microbars.

What is the actual sensitivity difference between these two mics? Before you say, "Who cares?" remember all you have to do is add 20 dB to the Mic #1 spec, if the Mic #1 impedance is 250 ohms:

dBm/10 microbar = dBV/microbar +20 dB = -70 +20 = -50 dBm.

So, Mic #1 has a sensitivity of -50 dBm, and Mic #2 has a sensitivity of -55 dBm. Microphone #1 has 5 dB more sensitivity than Microphone #2.

Whenever you compare microphone sensitivity specs, note carefully which reference SPLs are used, and convert them all to the same reference.

If you put a microphone in a 20 dB louder sound field, it produces 20 dB more signal voltage. For example, if 74 dB SPL in gives you -75 dBV out, then 94 dB SPL in gives you -55 dBV out. 150 dB SPL in gives you +1 dBV out, which is approximately line level! That's why you need so much input padding when you record a kick drum or other loud sources.

Summary

dB SPL = decibels of sound pressure level above 0 dB SPL.
dBV = dB ref. to 1 volt.
dBm = dB ref. to 1 milliwatt.
dBv or dBu = dB ref. to .775 volt.
dB in general = 10 log P1/P2 (power) or 20 log V1/V2 (voltage).

Balanced equipment, nominal line level at input or output = +4 dBm = 1.23 volts.

Unbalanced equipment, nominal line level at input or output = -10 dBV = .316 volt.

Difference in voltage level between +4 dBm and -10 dBV = 11.8 dB.

0 VU signal level = 0 dBm.
0 VU also means a zero reading on the VU meter.
0 VU on the meter corresponds to +8 dBm, +4 dBm, or -10 dBV level at the equipment output (for a steady tone).
0 VU record level = record level produced at "0 VU" meter reading.

1 pascal = 10 microbars = 10 dynes/cm = 94 dB SPL.
1 microbar = 1 dyne/cm = 74 DB SPL.
Microphone open-circuit sensitivity measured in dBV/pascal, dBV/microbar, or dBV/dyne/cm.
Microphone power level is measured in dBm/10 microbar, or dBm/10 dynes/cm, or dBm/pascal.

db = a magazine worth looking into if you're serious about audio.

For more detail on the decibel, I suggest the book *Sound System Engineering* by Don and Carolyn Davis, Howard W. Sams & Co., Inc., 4300 W. 62nd St., Indianapolis, Indiana 46268.

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This popular project is reprinted from the October 1980 issue.

Building A "HOT SPRINGS" Reverb

By Craig Anderton

WAIT! Don't turn the page. I know that simple spring reverb systems may not have the greatest reputation in the world, but this version uses a truly novel design technique. The result is a reverb system that offers an attractive combination of low cost and high performance.

You don't have to take my word for it, though. An engineer for a well-known manufacturer of effects boxes recently developed an all-electronic reverb system; part of his market research involved checking out the reverb market to see how his design compared to other currently available models. Since I felt he could be a little more objective about the "Hot Springs" reverb than I could be, I asked him to give it a listen. He was absolutely floored, and said it sounded better than anything else he had heard during his months of testing! I think you'll probably feel the same way after hearing it . . . but before we get into building, we need to examine just why the "Hot Springs" reverb (or "HS" reverb for short) is so different from the norm.

How Spring Reverbs Work

Let's begin by refreshing our memory as to how spring reverbs work in general *(see Figure 1)*. The spring connects to two transducers, one at the input and one at the output. Signals appearing at the output of the drive amp couple into the input transducer, which then takes this signal and couples it to a long spring; the signal is delayed as it travels down the spring. The output transducer picks up this delayed sound, and feeds it to a recovery amp which takes the extremely weak output of the reverb spring and amplifies it to a useable listening level.

So far, what we have described would only give a single "slapback" type of echo if it weren't for one very important fact: once the signal has reached the end of the spring, it bounces back along the spring towards the input, then reverses direction and bounces back towards the output again (contributing another echo), returns again towards the input, and so on until it eventually fades out. This creates the effect of multiple echoes and reflected sounds—just like you get in a large room. Also, there are several mechanical resonances in the spring itself that add peaks and dips in the response. This helps to simulate even more closely the properties of "real-world" reverb.

However, there are some problems (aren't there always!). The first is that

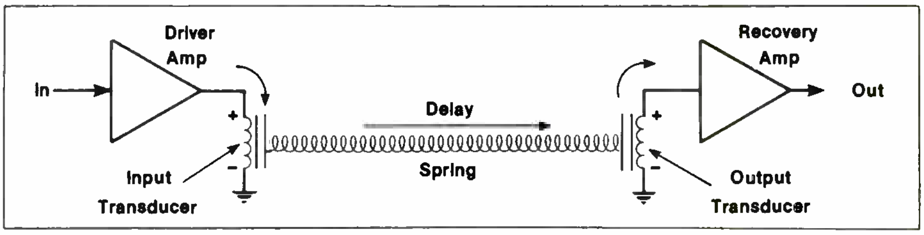

Figure 1

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the motion of the spring itself adds a certain type of sound to the audio signal, which produces the characteristic "boing" and "twang" of spring reverbs. The second problem is that if you just listen to the reverb output, you'll hear a mushy version of the "dry" sound along with the sounds created by the multiple reflections and echoes we mentioned earlier. The third problem is that the spring output is in the millivolt range, which is exceedingly weak. As a result, the recovery amp must run at a very high gain to bring this signal up to a useable level, and this contributes noise to the system. The final problem we'll discuss is that springs have an inherent bandwidth limitation, which means that there is no significant audio energy above approximately 5 kHz. This is why springs often sound bassy and boomy compared to a good, crisp plate system.

Solving these Problems

The HS design uses "hot rod" guitar pickup technology to overcome the above-mentioned problems. This is one of those situations where the solution seems so obvious you wonder why no one has thought of it before; but to the best of my knowledge, and several other people, the following represents a totally original approach . . . so *MR&M* readers, you heard it first.

The basic principle is to take two springs and connect the input and output transducers in a special way, as shown in *Figure 2.* The input transducers are connected in series and *out-of-phase;* the output transducers are connected in series but *in-phase.* As a result of the out-of-phase input connection, the original audio signal—as well as the "sproings" and "boings"—cancel each other out at the output, leaving mostly the multiple echoes and reflections. This neatly solves problems 1 and 2, and gives a very rich reverb sound. Additionally, the input transducers are driven by a constant-current source that provides equal drive for high and low frequencies. This gives the bright high end associated with plate systems, while de-emphasizing the muddy, bassy sound often encountered with some spring reverb designs. Finally, by connecting the output coils in phase and in series, we double the overall output level. This means that the recovery amp doesn't have to provide quite as much gain, thereby giving an improved signal-to-noise ratio.

You might wonder why the cancellation effect discussed above doesn't cancel the *entire* reverb signal. Luckily, although reverb springs are matched closely enough so that the "boings" and dry signal are mostly cancelled, there are enough differences in response (particularly in the high frequency regions) so that the subtler reverb sounds are left pretty much unaffected.

Musically speaking, we traditionally think of reverb as trying to simulate the sound someone sitting in the audience would hear at a concert. However, anyone who has played on the stage of a 2000 to 10,000 seat venue knows that reverb sounds quite different from the performer's perspective; it is this sound which the HS reverb simulates. Instead of hearing an ill-defined reverb mix of dry signal and hall acoustics (as you do in the audience), from center stage you hear the reverb coming back at you without any discernible dry signal. What this means in the studio is that the HS reverb sound never "steps on" the signal being reverberated, since it contains the multiple reflections and echoes associated with a good reverb sound *while excluding virtually any trace of the original signal.* This is highly desirable, since in practice the reverb signal is mixed in at a low level compared to the signal being reverberated. By cancelling the muddy sounding version of the dry signal that comes out of most spring reverbs, the overall sound is clean, crisp and well-defined instead of being boomy and sproingy. Vague terms, to be sure, but if you've worked with inexpensive spring reverb systems in the past I'm sure that all the above expressions will sound familiar.

Controls and Options

Actually, the only control for the reverb unit is a level-matching trim pot. At maximum sensitivity, signals greater than −10 dB will overload the driver amp. At minimum sensitivity, clipping does not occur until the input reaches +15 dB or greater. There is an additional clipping indicator LED that lets you know when the driver amp is being overloaded. Due to the high-frequency boosting action of this stage, clipping will occur sooner at higher frequencies than at lower frequencies.

Finding Parts

As mentioned in the last D.I.Y. ("do-it-yourself") Limiter article *[Modern Recording & Music, November 1979]*, whenever possible I try to line up a parts source that stocks all of the parts necessary to build a given circuit, as well as provide a repair service for wiring jobs that go astray. For this project, PAIA Electronics (1020 W. Wilshire Blvd., Oklahoma City, OK. 73116) is again providing this service—particularly because they stock the Accutronics model #1FB2B1D reverb springs which are used in this project. These springs were chosen for their low cost, small physical size, sound quality and ready availability from PAIA. As a result, all circuit components were selected with these springs in mind. While other springs may be used with this project, I cannot guarantee the quality of performance if substitutions are made. All other parts are commonly available and shouldn't be hard to find at all. If you decide to build the

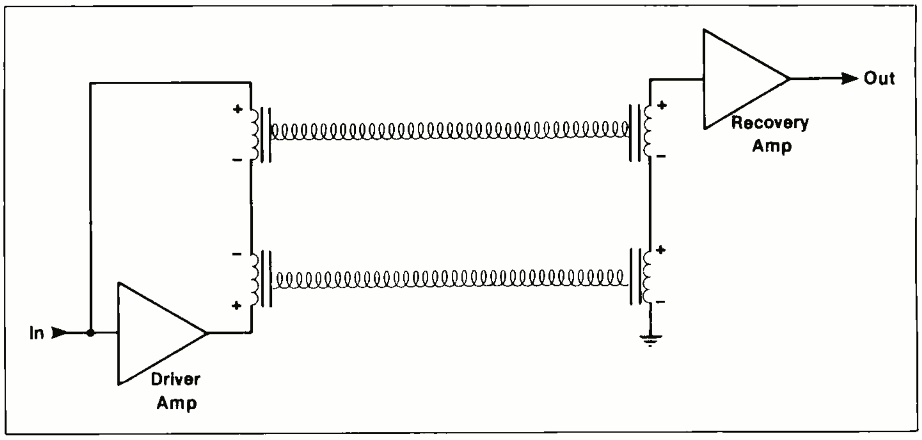

Figure 2

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project from scratch, however, I highly recommend that you follow the circuit board layout as closely as possible to prevent ground loops, hums, oscillations and other potential problems.

Preliminaries

While this is a fairly simple project, some aspects of it (such as modifying the reverb springs) require a bit of skill. So, I wouldn't recommend that beginners undertake this project unless they have successfully completed similar projects in the past. On the other hand, I said the same thing about the D.I.Y. Limiter and several beginners built it with no trouble at all. You are probably the best judge of your capabilities. However, there are some basics which *must* be observed, namely:

■ Use a low wattage (40 watts or less) soldering iron. *Do not use soldering guns!*

■ Use only rosin core solder designed for electronic work. Any kind of acid core solder, or use of flux, will ruin an otherwise good circuit board and some of the parts as well. Use the solder sparingly; don't blob it all over a connection, since that can cause shorts between adjacent circuit board traces.

■ The amount of heat used in soldering is very important. Too little heat can cause "cold" joints, where the solder's rosin is not sufficiently melted; this causes a high-resistance connection. On the other hand, too much heat will damage parts. I'd suggest holding the iron tip against the connection to be soldered for a few seconds, then feeding in a little bit of solder and continuing to apply heat until the solder flows freely over the connection. If the solder balls up around the connection, reheat it and feed in a little more solder.

■ Use an IC socket. This simplifies replacement should the IC ever fail; also, you don't have to worry about frying the part through incorrect soldering techniques *(see Figure 3a).*

■ Clean the copper side of the circuit board with steel wool to remove oxidation. A bright and shiny board contributes to successful soldering.

■ Note that electrolytic and tantalum capacitors are "polarized" components and have (+) and (−) marks, just like a battery. Like a battery, if you don't hook these parts up right the circuit won't function; so, the circuit board legend has a (+) symbol near the hold where the capacitor's (+) lead must go. LEDs and diodes are also polarized. Referring to *Figure 3b,* the LED symbol is an arrow pointing towards a bar. Generally, the bar (or cathode) end of the LED is designated with either a flat indentation in the case or a dot of paint. Diodes have a similar schematic symbol, and the bar end of the diode corresponds to a band painted on the diode itself.

■ Take your time and work carefully. Impatience is one of the biggest reasons why do-it-yourself projects fail.

■ No power supply is shown in the schematic. If you built the Limiter,

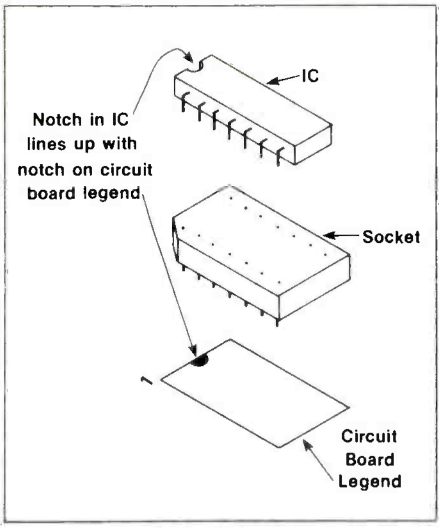

Figure 3a

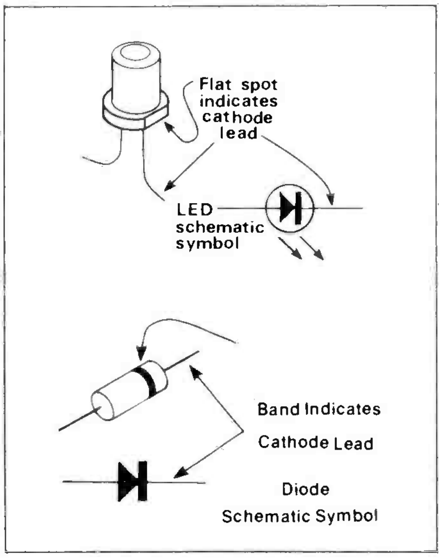

Figure 3b

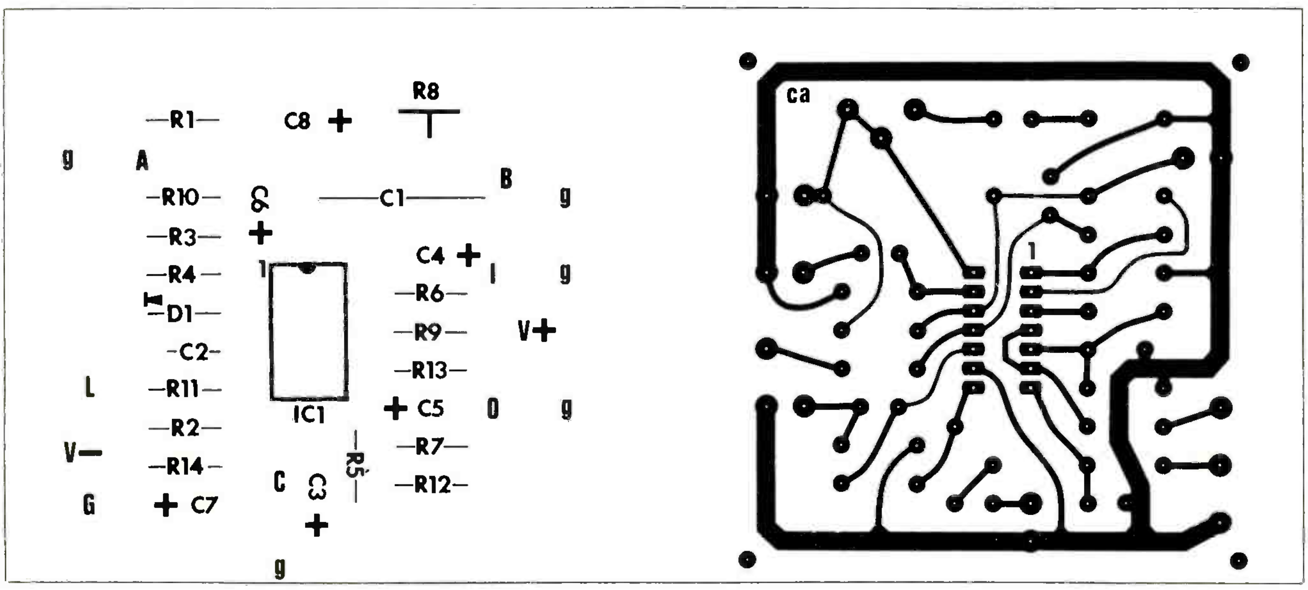

Figure 4

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you can use the same power supply; simply tap off another set of connections for the reverb unit and you're ready to go. Otherwise, you can use any +/−15 V bipolar supply such as the PAIA #4771 or the HK-116 from Bill Godbout Electronics (P.O. Box 2355, Oakland Airport, CA. 94614).

Space prohibits us from going into all possible aspects of electronic construction. If you'd like to find out more about this topic, refer to my *Electronic Projects for Musicians* book (published by Music Sales, 33 West 60th Street, NY, NY 10023) *[Craig . . . is this a blatant plug for your book!?—Ed.].* It contains complete information on finding parts, soldering, packaging projects, labelling, etc.

CONSTRUCTION: There are four distinct phases to construction: 1) loading and soldering the circuit board; 2) modifying the reverb springs; 3) packaging the springs and circuit board in a suitable chassis; and 4) connecting the circuit board to the power supply and springs. We will deal with each one in order.

SOLDERING the CIRCUIT BOARD: Referring to *Figure 4* (the component side of the board) and the parts list, solder the various components in place. Start with the resistors first, then the IC socket, capacitors and trimpot. Check that all solder connections are well made, then proceed to the next section.

MODIFYING the REVERB SPRINGS: In order to do the various in-phase and out-of-phase tricks mentioned in the beginning, we have to modify the wiring of the two reverb springs. This is probably the most complex part of the project, so pay careful attention to the following instructions.

Begin by placing the two reverb units side-by-side with the springs facing up, as shown in *Figure 5;* note that the two jacks are facing to the left. The input jacks are towards the top of this figure, and the output jacks towards the bottom. Since the input jacks are easiest to wire up, we'll do them first.

Referring to *Figure 6* (which shows the modified wiring), disconnect the black wire attached to each reverb spring's input transducer from the associated ground lug of the input jack. Next, note that there are some little springs that hold the spring plate to the case, and that these springs hook on to a hole in the side of the reverb spring case. Now connect a small piece of thin gauge insulated wire to the black lead of the left-most reverb unit,

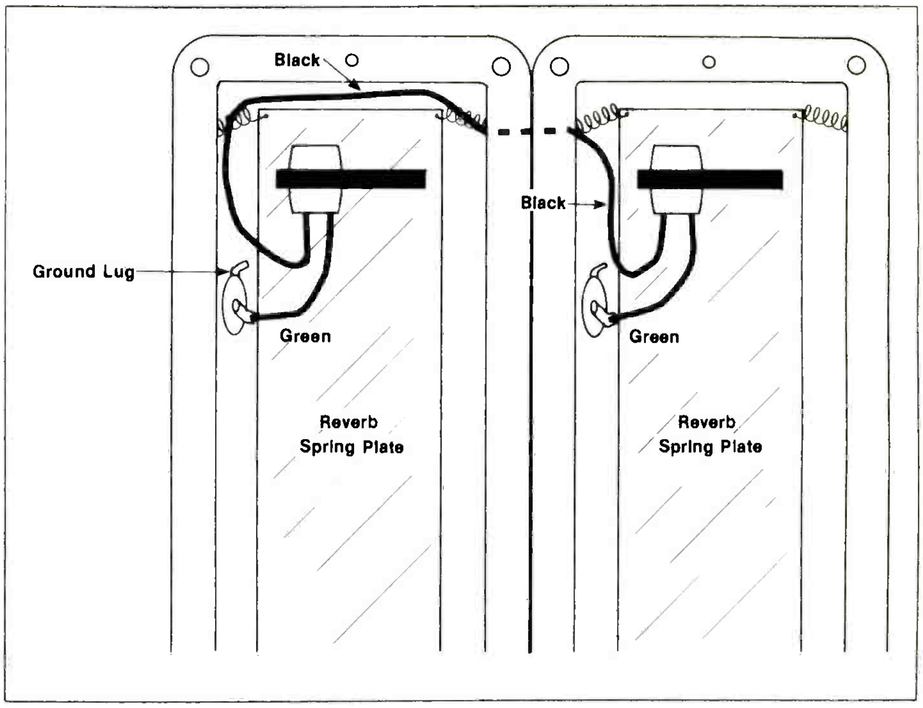

Figure 6

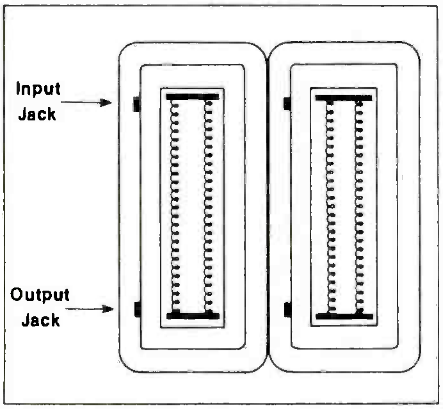

Figure 5

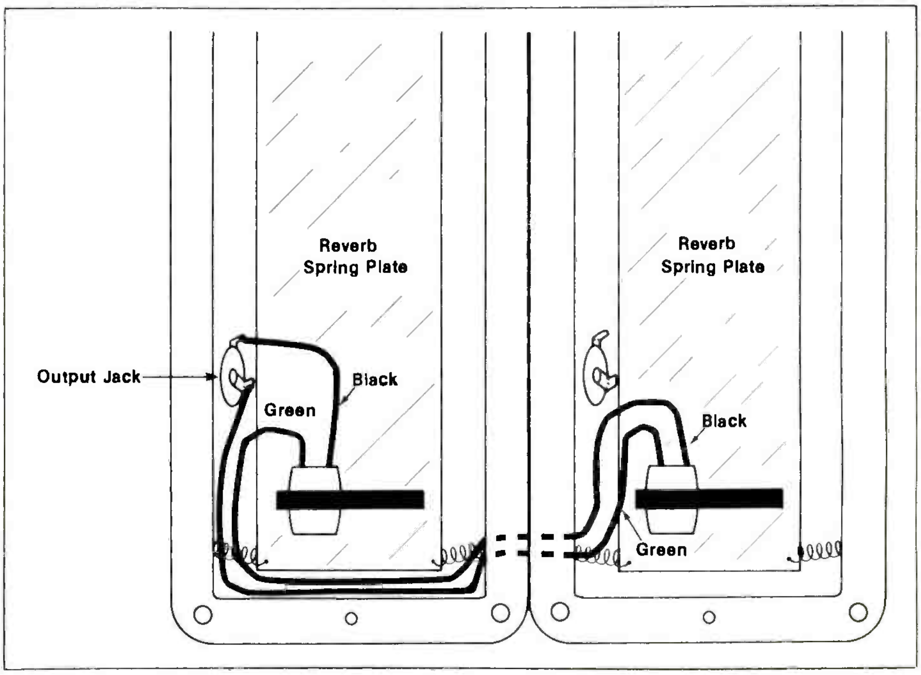

Figure 7

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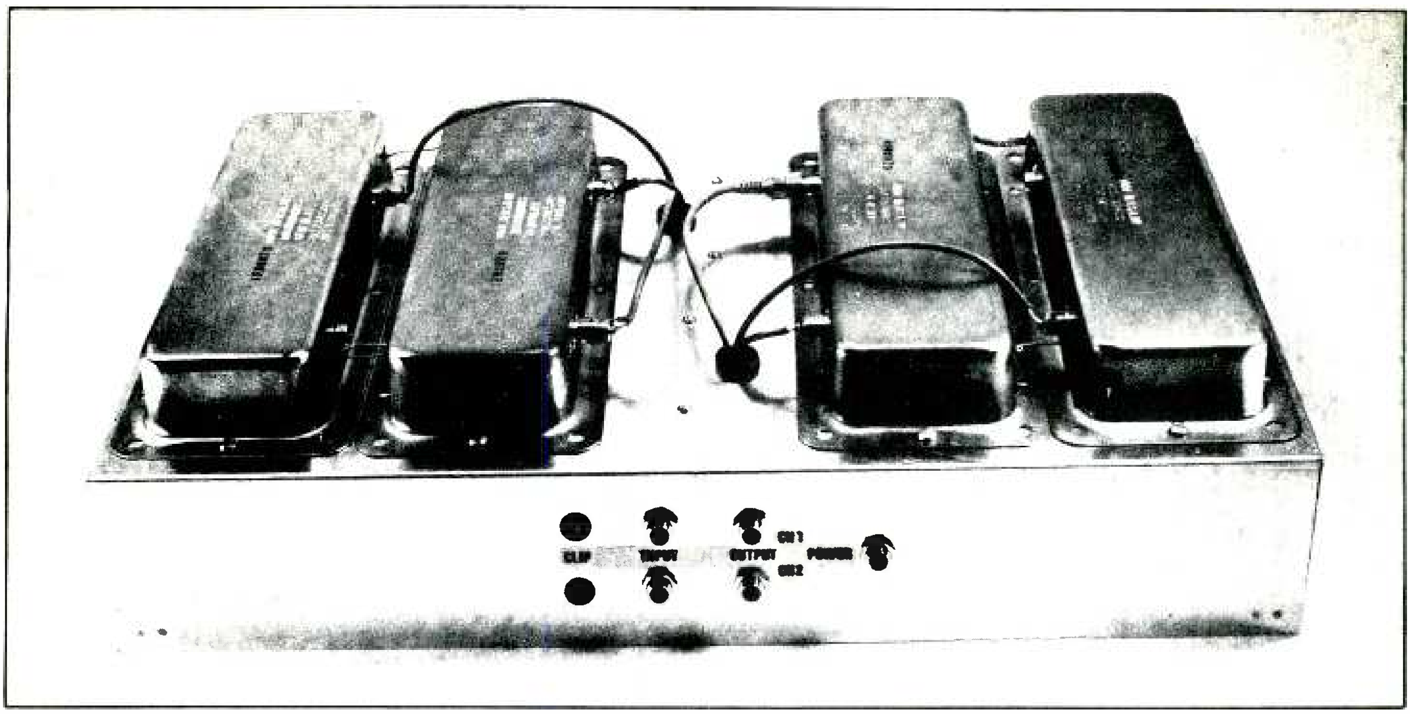


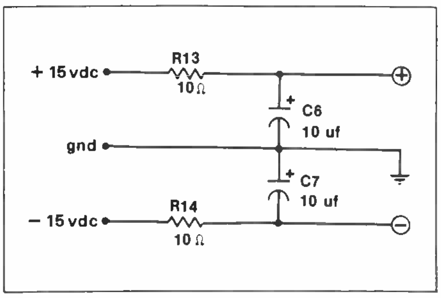
Figure 8

run it around the inside of the case as shown, and run it through the holes in both cases where the little springs hook in. Then, after you've gotten this wire inside the reverb unit on the right, connect it to the black wire coming from the remaining transducer. Finally, use a thin piece of electrician's tape to insulate the connection between the transducer leads and the added "jumper" lead. Check with *Figure 6* again to make sure everything is connected correctly, and that the extra length of wire does not interfere with the free motion of the reverb spring plate.

Figure 7 shows a detail of the modified output jack wiring. In this case, disconnect the green wire from the left transducer and the black *and* green wires from the right transducer from their associated phono jacks. Connect the green wire from the left transducer to a short length of thin gauge insulated wire, and run this into the right reverb case through the case holes (like the ones mentioned above). This wire should then connect to the black wire from the right transducer. Finally, the green wire from the right transducer should connect to a wire that again runs through the two holes used for routing the last wire, and ends up connecting to the "hot" terminal of the output jack mounted on the left reverb unit. Look carefully at *Figure 7* to check that all is well. It is important that the wires not interfere with the motion of the springs or the plate to which they connect. Make a mark on the case near the output jack so that you don't forget which one is wired to the springs.

(If you feel ambitious, the two wires connected to the output transducers can be shielded. However, you'll have a hard time finding shielded wire that's skinny enough to be comfortably routed as shown in the diagram—luckily shielding isn't absolutely necessary.)

Now that the springs have been modified, it's time to find a suitable enclosure to hold the springs and circuit board. *Figure 8* is a photo of the case I used, which is a general purpose aluminum chassis sold by electronics supply houses. Since I wanted stereo reverb, I used four springs (two for each channel). These springs should be mounted as shown; do not mount them upside down or sideways, as they don't sound right that way. In my particular case, I mounted the two circuit boards for the two channels inside the box, and ran the connections from the boards to the springs through a few holes drilled in the chassis. The input/output jacks and LEDs mount on the front of the box as shown.



Power supply connection

Connecting It All Together

Now we come to the last stage. Run a shielded cable from pad I on the board to the input jack; connect the shield at the board end only. Note that there is a pad next to pad I (pad "g") where you can connect the shield. Next, run a shielded cable from pad O to the output jack; again, connect the shield at the board end to the pad "g" next to pad O. All future steps involving shielded cable should have the shields connect to the nearest pad "g" on the board. Do not confuse these with point "G," whose use will be covered later.

Now connect a length of shielded cable to point A, and terminate it in an RCA phono jack. This wire should be long enough to reach either reverb spring input jack. The shield should *not* connect to the plug's ground, but just to pad "g" on the board. Plug this ungrounded phono jack into the reverb spring input. Then, in a similar fashion, connect a piece of shielded cable to point B, with its shield con-

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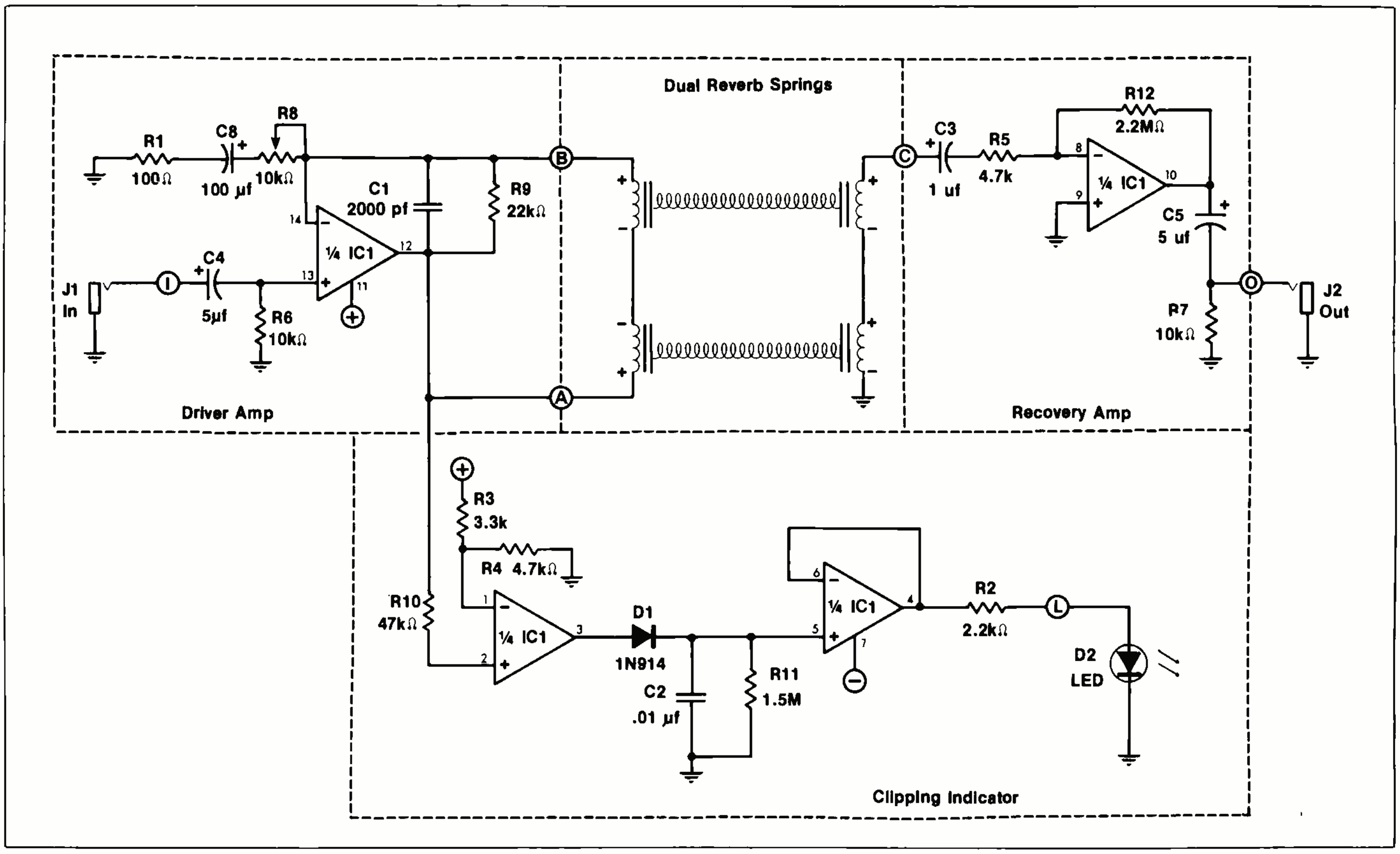

Figure 9

nected to the "g" pad near B. Again, check that the shield does *not* connect to the plug's ground. The ungrounded phono plug connected to this wire should plug into the remaining reverb spring input.

Our final piece of shielded cable connects to point C, with (you guessed it) the ground connecting to the nearest point "g." This wire should terminate in an RCA phono plug; *but this time, make sure that you do connect the shield to the plug's ground,* then plug into the reverb spring output jack you marked in an earlier step.

O.K., now we've connected the springs to the circuit board and the circuit board to the input and output jacks. Our final task is to hookup the LED and power connections. Run a wire from pad L to the anode of the indicator LED; run a wire from the cathode of the LED to a convenient ground point, such as the ground tab of the output jack. For power, connect pad G to the ground tab of the input jack, then connect the +15 V line from your power supply to pad V+, the −15 V line from your supply to pad V− and the ground line from your supply to the chassis ground or input jack ground tab. In my version, I used a stereo jack

SPECIFICATIONS

Maximum input before clipping: −10 dB (maximum sensitivity), +15 dB (minimum sensitivity)
Input impedance: 10 k (may be changed to 100 k by replacing R6 with a 100 k resistor and C4 with a .22 uF capacitor)
Output impedance: Less than 1 k
Current consumption: +7 mA, −7 mA
Signal-to-noise ratio (peak output compared to residual noise): greater than 63 dB
Frequency response of reverb signal: (please note, due to the various resonances and uneven response desirable in a reverb unit, it is difficult to give accurate response figures. The *Figure 10* graph is an attempt to average the response to give a meaningful composite figure).

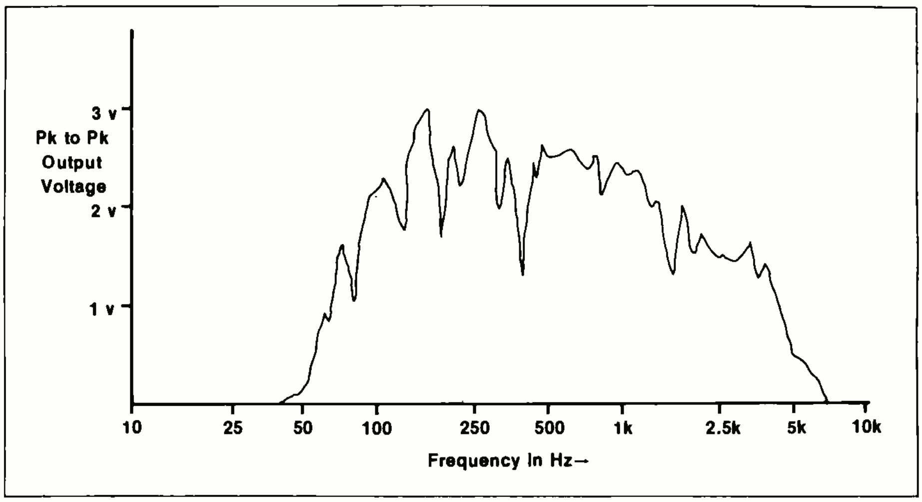

Figure 10

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"HOT SPRINGS" REVERB PARTS LIST

Resistors

All resistors are ¼-watt, 10% tolerance unless noted. 5% tolerance resistors are preferred.

R1	100 Ohms
R2	2.2 k (2 k 2 metric)
R3	3.3 k (3 k 3 metric)
R4, R5	4.7 k (4 k 7 metric)
R6, R7	10 k
R8	10 k trim pot
R9	22 k
R10	47 k
R11	1.5 M (1 M 5 metric)
R12	2.2 M (2 M 2 metric)
R13, R14	10 Ohms

Capacitors

All capacitors are rated at 15 or more working volts.

C1	2000 pF or 2200 pF (2 nF metric) polystyrene
C2	.01 uF (10 nF metric) disc ceramic
C3	1 uF electrolytic or tantalum
C4, C5	5 uF electrolytic or tantalum
C6, C7	5 to 50 uF electrolytic or tantalum
C8	100 uF electrolytic or tantalum

Semiconductors

D1	1N914 or equivalent silicon diode
D2	Red LED
IC1	RC4136 (Raytheon) or XR4136 (Exar) quad op-amp

Other Parts

J1, J2	Open circuit ¼-inch phone jacks or RCA phono jacks (depends on your particular setup)
Misc.	14 pin, IC socket, Accutronics #1FB2B1D reverb spring, circuit board, solder, case, wire, etc.

(Note: a parts kit containing the above mentioned items, less case and solder, is available from PAIA Electronics for $59.95; specify #6740-K. The circuit board is available for $7.95; specify #6740-PC. The reverb springs are also available for $22.95 each. All prices are postpaid in the USA.)

on the front for my power supply wiring; this enables me to use a stereo cord to plug the reverb unit into a multiple-outlet power supply.

One more thing: If for some reason you mount the reverb springs on a non-conductive material (e.g., plastic), run a wire from each reverb spring case to ground. It is important that the spring cases be grounded to keep hum to a minimum.

Testing Time

Connect the output of the reverb system to a suitable monitor amp (with the volume turned down!), then patch an instrument, tape track, or similar signal source into the reverb input and apply power. Turn up the monitor; you should hear the reverberated sound. Now observe the indicator LED. If it doesn't glow very much, *increase* the sensitivity trim pot so that it flashes on signal peaks in order to avoid excessive noise. If, on the other hand, the LED flashes a lot, *decrease* the sensitivity to avoid distortion. The setting of this trim pot is rather important, so don't be afraid to experiment. If you find that the reverb output is too noisy, make sure that the sensitivity control is set properly in order to give the maximum possible level to the springs short of distortion.

How It Works

Referring to the schematic *(Figure 9)*, IC1A is the driver amp. Capacitor C1 tunes the reverb for a response peak at about 5 kHz, while R9 sets a ceiling on the maximum amount of gain generated by this stage. IC1C and IC1D tap off the output of this stage and comprise a simple clipping indicator. If the signal appearing at IC1's output exceeds the threshold set by R3/R4, then IC1C turns on and charges C2 through D1. C2 acts as a pulse stretcher to catch short duration transients, with the decay time being set by R11. IC1D simply buffers this cap and drives the clipping indicator LED.

Signals appearing at the spring output drive IC1B, the high-gain recovery amp stage. The 2.2 M feedback resistor is kind of extreme, but of all the configurations tested this one gave the lowest overall noise figure. R5 is a low enough value to load down the springs just a tiny bit, which reduces excessive high frequency response that would otherwise add a kind of "tinniness" to the sound. You can substitute a 10 k resistor for R5 if you'd like to trade off more noise for extended bandwidth, but I think 4.7 k gives the best overall results.

By the way, if the reverb unit has poor lead layout or ground loop problems (which it shouldn't if you followed the instructions carefully), it's possible that IC1B will oscillate. To avoid starting over from scratch, you can fix the problem by adding a 10 to 20 pF capacitor in parallel with R12. However, this should not be necessary if you grounded your shielded cables correctly and used the circuit board layout shown in *Figure 4*.

In Conclusion

I hope you get as excited about this reverb unit as I am; I think it sounds real good, and am happy to be able to share it with the recording fans who devour *MR&M* each month. If you have any questions about the reverb's operation, or run into difficulty, be sure to write so that we can cover any problems in future issues of *MR&M*.

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bob buontempo

Making A Plate Reverb Unit

This interesting project is reprinted from the May 1983 issue.

The weakest link in a small recording studio (equipment-wise, not counting the engineer or musicians!) is usually the reverb unit. This is what sometimes makes the difference between the sound of a "demo" and a "master."

This article presents plans for making a plate reverb unit, not requiring any electronics other than a mixer and headphone amp, already owned by the studio and/or reader. Utilizing the headphone amp, which is usually never used during mix-down, gives the amp a dual function in the studio and makes it very cost effective. The cost will be between $300 and $500, 1/10 to 1/30 the $1,000 to $9,000+ for commercially available units. The following will describe how to find and evaluate materials needed; how to physically construct the frame; mount and tune the plate; fit the driver and pickups; dampening of the plate; and "tricks" and techniques for enhancing plate sound.

Diagrams, photos, and parts list are included. The plate design will constantly be compared to commercial plate designs for evaluation, technical, and explanatory purposes. A kit providing a pre-constructed frame; selected, cut, reinforced, and drilled plate; mounting hardware; driver, and tuning cassette tape, as well as the driver and tuning tape alone is offered. This will help facilitate the building of a unit by eliminating the hardest part of the project; evaluating and locating materials and having the custom metal working prepared. We have used the plate that we constructed and based this article on in our studio (Home Grown Studios) for over five years. It has been used on records and tapes featuring Kinky Friedman, Dr. John, Max Romeo, the Late Nite Band, the Stanky Brown Band, and sax great Phil Woods. Also on many other projects that included members of bands such as Blue Oyster Cult, Billy Squier, the Dixie Dreggs, David Johansen, Ian Hunter, and the Clarence Clemmons Band, among others.

Almost everyone into recording is familiar with spring reverbs, or at least their sound. Most "low end" or

PARTS LIST

1 sheet of steel, 3-ft. by 6-ft. ("304" 26 gauge stainless steel recommended).

Tubular steel or angle iron for frame.

1 Acoustic 2000 driver (or alternate driver).

10 fiberglass (preferred) or steel yokes (or hooks) with appropriate nuts, bolts, metal and steel washers.

Rubber straps or shock absorber bushings for suspension of frame in case.

Wood for case.

1 damping plate and absorbent material with handle and hardware.

1 or 2 pickups (of choice) or recommended units.

1 "passive drive signal response shaper."

1 tuning/EQ cassette.

For information on drivers, pickups, response shaper, tuning/EQ tape and partially or completely assembled kit, write to:

Bob Buontempo
P.O. Box 531
Cranford, NJ 07016

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"semi-pro" commercially available reverb units are based on a spring principle. So are most musical instrument amps or accessories with reverb. The "spring sound" can range from excellent to "under water" depending on the unit and the way it is used.

The reason spring units sound the way they do is because that is exactly what they are—springs—similar to a lighter version of what you find on your screen door. There are usually several rows of them, with possibly two or three strung in a series, and like the springs on your screen door, when plucked, they will "twang" or "boing." A spring reverb unit is almost the same but instead of being plucked, the springs are excited on one end by a driver and picked up on the other end by a pickup...and so is the resultant "twang" and "boing," especially on transient material.

Although some designers have tried "tricks" to smooth out their sound with excellent results (Craig Anderton's "Hot Springs"), they still have spring characteristics inherent in their sound, as well as a limited bandwidth, especially in the high frequencies (8 kHz+).

Plate reverb, however, has none of these drawbacks; it can go from sounding like a true concert hall to an oil drum being banged with an ax in the subway, again depending on its application and who's using it.

What is a plate exactly? A plate is typically a one by two meter (39½ by 79½-inches) sheet of steel suspended in a tubular steel frame. The plate, in theory, simulates a large concert hall or church with a decay time (the length of time required for the intensity of the reverb to diminish by 60 dB) of approximately five seconds at approximately 500 Hz. A driver, acting as the instrument or singer, excites the plate, and, as the sound waves travel through it, they reverberate as they would in a hall. (See *Figure 1*.) They are then picked up by one or two contact mics, and added to the dry signal at the mixer. The signal retains its full frequency response, and transients do not "twang" or "boing" but behave much as they would in the room, sounding smooth and natural. As an additional feature, incorporating a damping plate to change the decay time of the reverberated signal can be included in the design.

It was at the Broadcast Technical Institute in Nurenburg, and later at the Institute for Broadcast Engineering in Hamburg, West Germany, that the first reverberation plate using these principles was developed. EMT (in Germany) patented and made the sole available units until the patents ran out a few years ago. Several American and foreign companies have come out with new units that are basically modified and updated EMTs. The plans presented here are to build a hybrid unit that can be optimized to the design of any of the commercial units you favor.

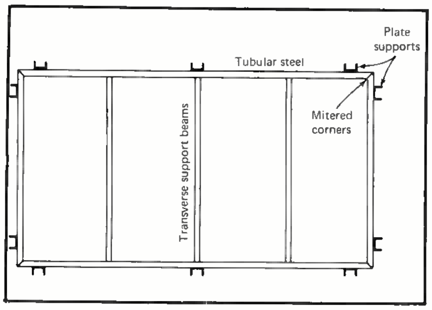

Figure 1. Placement of driver and pickups and effect of sound waves on plate.

Construction of the Unit

As mentioned in the introduction, the design of this unit will incorporate your mixer and cue (headphone) system as all the electronics that are required, so we will mostly concentrate on the construction of the physical unit—frame, plate, driver, and pickups.

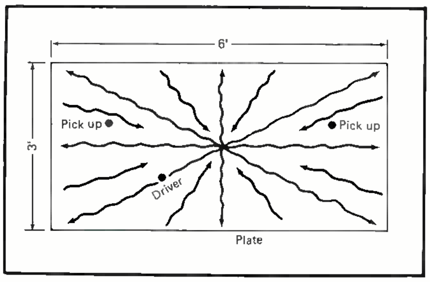

Figure 2. The tubular steel frame is reinforced with three transverse support beams.

Selection of Plate Materials

This is probably the most critical of all the steps involved in the process so be careful in your choices. The plate is actually the "instrument" used for the reverb, so it should be chosen as if it were a fine acoustic instrument—guitar, piano, sax, etc. EMT uses a one meter by two meter cold rolled steel plate approximately 1/64 of an inch thick. Lawson, who manufactures "the Plate" uses basically the same thing. On the other hand, some manufacturers use stainless steel. The Ecoplate by Studio Technologies uses approximately the same gauge in stainless, as does Audi-ence from Lake Dallas, Texas. DB Cassette of Sweden, who manufactures the Stocktronics Plate, use a stretched, hardened piece of cold rolled stainless. The Reverb-Tron plate gives the user a choice of either type. Most other companies use stainless steel. The debate about what kind of steel to use is totally subjective. Reasons claimed for using stainless include consistency of steel quality, high density, more high end response, and rust and tarnish resistance, while cold rolled steel users claim smoother, more natural sounding reverb, and a less metallic decay. Only you can decide what sound you prefer.

We recommend using 26 gauge stainless steel. Almost *any* piece you use will sound fine; the consistency is *that* accurate. It gives a bright contemporary sound to your mix. Specify a "304" alloy number and you can be reasonably assured of getting what you want. Befriend your local steel warehouse owner, bring two associates, and prepare to listen. Most steel sheets come 3-ft. wide and this is close enough to one meter for our purposes. The length, however, is usually eight to twelve-feet long and cutting charges for shortening it to six-feet might be added to the price of the steel. Also, some place have minimum order amounts, so try to buy your plate and frame materials from the same source to save added expense.

If the owner of the shop will allow, (and it is worth the price of a healthy tip to have him help you out), have your two friends hold the sheet of steel you are considering upright as tight and still as possible, so it doesn't "thunder," and tap it in the center with a key. Listen for a "sizzle" and long decay in the high frequency as opposed to a "clangy" sound. The delicacy and length of the high frequency are what you are really after since the bottom and mids can be dealt with more successfully by tensioning, EQing, filtering and damping. Try several pieces of different types until you find what you are after. Be selective and take as much time as allowed, because this is the heart of your system and you must be happy with it.

Including the cutting, the steel sheet (depending on the type you have chosen) should run between $50 and $100. Reinforcing the corners by spot welding a triangular piece of steel on each one is the recommended procedure. For corner cutting by the cost

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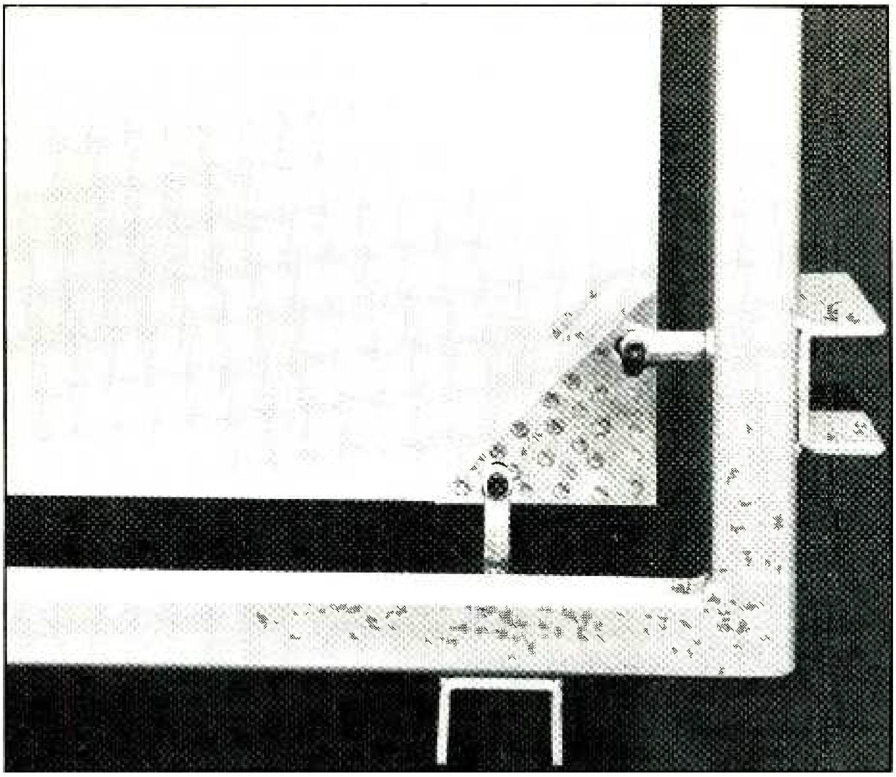

Figure 3. Corner detail, showing triangular reinforcement welded in place.

conscious it is not totally necessary (since it could run $25 to $50 additionally), but it really should be done if at all possible, because the plate will be put under heavy tension and holes drilled in the corners as the next step in the plate preparation procedure. The holes should be drilled after the frame is completed so a more "custom" fit is accomplished. If you don't use stainless steel, apply a very thin coat of the lightest machine oil you can find to the plate to prevent rust and corrosion.

The Frame

The frame is simply 1 to 1½-inch tubular steel—either round or rectangular—shaped in a rectangle and welded together at the corners (preferably mitered) and reinforced by three equally spaced beams (See *Figure 2*) (or four, two in the center, if an alternate driver is used—but more on this later). Angle irons could be used as a super economy measure (as cheaper units do) but tubular steel is the more popular and recommended method. Additionally, on both sides of each of the four corners (eight all together) flat pieces or slats of steel, channeled for extra strength, are welded on—extending out 1½ to 2-inches beyond the frame 1 to 2 inches from each corner. (See *Figure 3.*) Add two more supports to the center (top and bottom). This will virtually eliminate tuning problems by stopping the beating and rumble of the plate by keeping the center tight. The kit comes with two additional slats in the centers of the longest sides of the frame. These help even out the tension on the plate, stop the "floppiness" of its center, and "tighten up" the sound. It is recommended that you add them too, so they will be available for you to try if you desire. Holes will also be drilled in the center of each of the slats. (See *Figure 4.*) To determine the exact placement of these holes in the slats and in the plate, as well as the exact measurement of the length of the tubular steel for the frame, you must make sure the plate and the frame "line up" together. If your plate is 3-ft. by 6-ft., make the *inside* measurement of the frame 1 to 1½-inches larger than the outside of the plate. Then lay the plate inside the frame. Mark the ten spots where the holes will be drilled in the plate, then align and mark where the ten slats will be welded to the frame, and the centers of the slats where the holes will be drilled. When all the holes are drilled and the slats welded, paint the frame to stop rust and corrosion. The next step is to suspend the plate in the frame.

EMT uses spring "clips" that hold the plate in place which are also used to determine tensioning. Some think these are weak, and they often snap, and one of the changes made by most of the newer plate manufacturers is the replacement of these with stronger, heavier clips or hooks. Ecoplate uses clips similar to those that secure fiber straps used on packages. Audi-ence uses turn-buckles, and the other companies use similar methods. Simple, tempered, hardened steel hooks, threaded on their shafts can also be used, which can be for this design. However, we found it easier and the tension more even when fiberglass yokes with bolts through them are used. (See *Figure 5.*) The kit includes "clevis yokes" with bolts and pins for the highest quality and ease of tensioning. If the yoke is fiberglass and a hard rubber washer and a metal washer are used, the plate and the frame can be totally isolated as far as direct metal to metal contact. To suspend the plate (you will probably need help with this), hook the hooks through the holes in the plate. Slip the shaft of the hook through the holes in the slats. Thread the washers and a nut of the correct size on the hook shafts and tighten all nuts hand

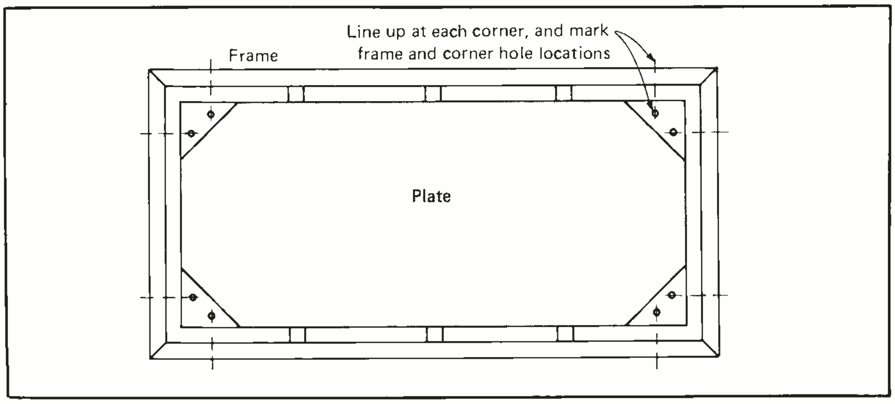

Figure 4. With the plate carefully positioned on the frame, both the frame and the plate may be marked so the holes will be properly aligned with the hooks.

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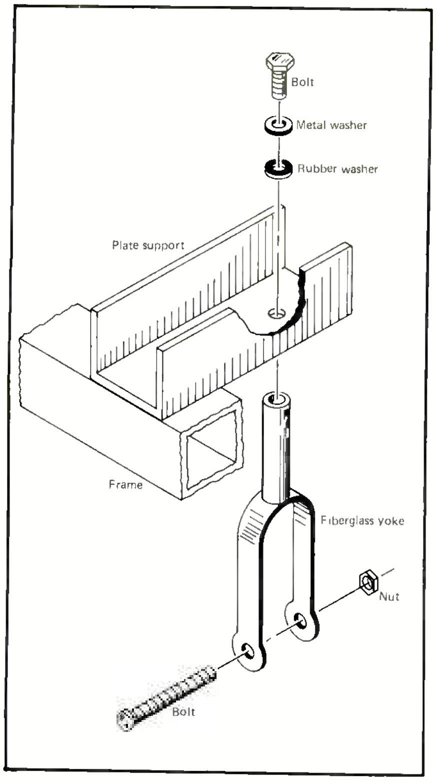

Figure 5. Detail showing correct positioning of suspension yoke.

tight. The plate will be suspended in the frame. Now comes another subjective and fun part of the project—mounting the driver and pickups and tuning the plate.

The Driver

EMT, Ecoplate, and Lawson all use similar drivers. A bullet shaped metal moving coil "slug" is screwed into the center of the plate. Two wires carrying the signal go to the coil and it is suspended in a large, heavy, circular magnet. (See *Figure 6*.) It is important in this design to be sure the moving coil does not rub or touch against the sides of the magnet, which is why the frames of these units have *two* center reinforcements

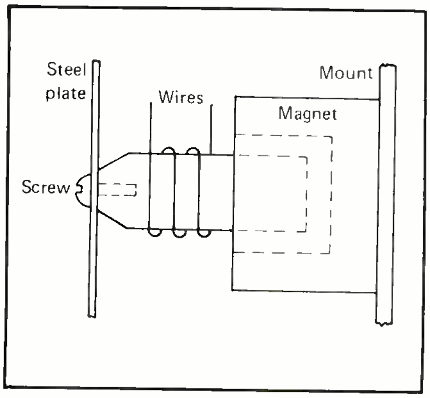

Figure 6. Driver detail.

instead of one as in this design. This enables the mounting of the magnet to be adjusted in all directions. The coil and magnet are aligned using a plastic alignment disk on EMTs. The procedure is delicate, and transporting the unit can misalign the driver/magnet assembly.

Stocktronics and Audi-ence use a wire rod attached to the plate on one end and to the voice coil of a speaker on the other. The Stocktronics unit can be moved with no realignment since there is plenty of "play" in the movement of the rod, and this is restricted to within limits by a rubber guide.

The system we will use is similar to both but unique unto itself. It is also one of the main reasons that this plate can be built so reasonably. This design uses an Acoustic 2000 driver—similar to what used to be offered as a "coneless speaker" several years ago. Whatever the driver is attached to becomes the "sounding board" and vibrates enough to reproduce sound. Therefore, if screwed into a wall or door, their surfaces would become a "speaker." The Acoustic 2000 is an improved version of the old "coneless speaker." It offers virtually flat, distortion-free, frequency response from 30-20 kHz. It can safely handle 35 watts RMS and 100 watt peaks, and has a built-in crossover, with a replaceable capacitor to change the crossover frequency. Best of all it is simple to use and to install and is very reasonably priced.

To install the driver, simply drill a small hole, the size for the screw on the driver, off the center of the plate 2½-in. to 6-in. off to one side of the center of the beam of the frame. The exact location of the driver is subjective, a matter of experimentation, and dependent on the individual piece of steel. The dead center of the plate is the worst spot for exciting standing waves, and an off-center placement gives the best results. When your pickups are attached to the plate and you are going for your optimum sound, try the driver in a few places by holding it against the steel before finally deciding on a permanent location by drilling the mounting hole. Joe Errico, who helped me develop and improve the plate over the past two years has found a "sweet spot" or perfect location for consistent sound in the driver placement. I call it the "J" spot. To find the "J" spot on your plate, measure 24-in. from the left side of the plate and 20-in. from the top. Drill it here. Place the driver in the hole on the side of the plate, with the frame reinforcements toward you. Thread the washer and nut on the screw of the driver and tighten until it is secure and won't rattle. Attach a speaker cable to the two terminals on the driver. Be neat and run the cable down the reinforcement with "ty-raps" or tape. Move the plate into your studio. Make sure the plate is standing upright. Connect the other end of the speaker cable attached to the driver to the output of your cue (headphone) amp—(or one channel of it if it is a stereo cue system). Put on a tape with a steady snare drum track or a constant vocal track. Send only the selected track to your cue and, voila! The signal will be heard on the plate. Assuming it is the snare track, what you should hear is thunderous snare similar to "Bridge Over Troubled Water" or "The Boxer" by Simon and Garfunkel (although I think they used an elevator shaft for their reverb chamber). But anyway, congratulations! You have a working plate reverb! Now comes the *real* fun part—using your opinions, taste, ego, etc., to get it to sound the way you want it to. This will require your choice of pickups as well as tuning and EQ.

Pickups

All the commercial units use Piezoelectric pickups or accelerometers. These are basically contact mic/pickups and are available from dozens of manufacturers. As a matter of fact, you probably already own one, or at least know someone who does. Some examples of recently available units are Barcus-Berry "Hot Dots," Frap, Shadow Pickups, DiMarzio, Adamas, etc. Some pickups need no preamp and can be plugged directly into the echo return(s) on your mixer. Some have their own preamps, but these tend to be rather noisy. An MXR Micro-amp or similar FET or DOD Bi-FET preamp would be a good substitute if needed because it uses a low noise, high slew FET for excellent quality. You can also return the output of the pickups through two mic inputs on your mixer if you have any modules free, using a direct box or transformer if necessary. These are variables that you must work out depending on the mixer you own and the pickups you use, as well as other variables in the equipment itself (impedance matching, levels, etc.). Try as many pickups as you can borrow until you find one

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that you like and that easily interfaces with your mixer. We have found the ideal pickups custom made for us in Europe. They have extremely high output and exceptionally low noise. They are very well shielded, and have a nice, bright, clean sound. (Information is at the end of this article.) If you do use any preamps, or other electronics (such as a separate amp instead of your headphone amp) they can all be mounted together, either in a rack or on the case of the unit—improvise whatever is best and most convenient for you. For a mono reverb place the pickup near one of the side frame reinforcements. Experiment by moving the pickup around up, down, on both sides of the reinforcement, until you find the spot you like. The "J" spot for pickup placement is 10-12-in. from the top of the plate and 10-12-in. from the side. This spot is the same for both the left and right sides. Then secure the pickup there by epoxy, wax, putty, or whatever the pickup manufacturer recommends. We recommend crazy glue or some type of super glue. Just use a drop and hold in place until secure. To easily remove the pickup, hold it with pliers and twist it off. Using this type of glue instead of a putty type will give a cleaner, clearer and brighter top end. Run the pickup wire down the reinforcement again using "ty-raps" or tape. For a stereo unit do the exact same thing on the other side. Pan one left and the other right. The differences in the sound waves reaching the pickups, as well as their phase, frequency response, and the time factors will give a very nice diffused stereo effect from a mono send. Most plates derive their stereo effect in this way.

Tuning the Plate

With the pickups hooked up, the plate itself now comes into focus for tuning. In theory, think of the plate as similar to a drum head—the tighter it is, the higher the pitch. Also, correct tuning means all the lugs are equally tensioned. So start by holding the yokes suspending the plate in the frame, securely, with a pair of "vice grips" or similar pliers, and tighten the nut of that yoke with a ratchet wrench. Do this evenly around all ten yokes. You can use a torque wrench, if that helps. How do you know when the plate is tuned? Good question. You don't really, every manufacturer uses their own method for tuning. EMT ships units pre-tuned except for four nuts which are supposed to be retightened by exactly ¼ turn when installed, but some independent EMT servicemen will tell you to tighten the plate until a spring suspension clip breaks, and then replace it and tighten until ⅛ of a turn before it breaks again! Lawson ships its units pre-tuned no adjustment necessary as does Audi-ence. Ecoplate, like EMT has you tighten four

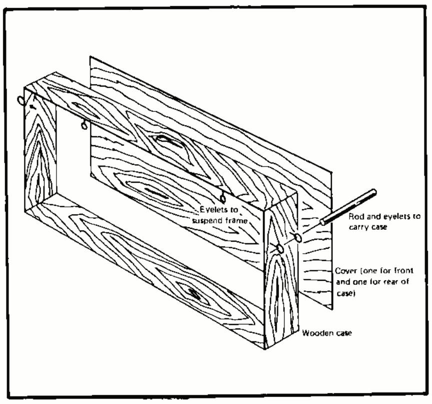

Figure 7. Drawing of frame with eyelets to suspend plate.

nuts ¼ turn, but also supplies a spring gauge and specifies pushing the gauge against the plate at all tensioning points until there is approximately the same pressure at each point. You could do the same with a spring scale or "fish scale." As a total contrast, Stocktronics uses no tuning at all. In fact, their plate is simply suspended by six springs in a very light aluminum frame, claiming the steel is pre-stretched/tuned during manufacturing. The Ecoplate III takes this approach also. Which method is correct? Any/all/none, depending on your point of view. One thing is certain though, if you like the way it sounds, it's right for you. So that's how I suggest tuning, by ear. Remember, usually the tighter the plate, the more highs/less bottom. It is also usually better to over-tension than under-tension. Also, listen for "flutters" or "beats" (like two slightly out of tune guitar strings) on the decay of the reverb and even out the tension until they disappear. EMT warns about the "oil can effect," a very metallic sound that is heard on an obviously out of tune plate. What I suggest is to find an existing plate you like the sound of in a studio around where you live. Rent an hour of time from them, bring a tape of various tracks—snare alone, drum kit, congas, tambourine, voices, piano, etc., and run it through their plate. Record the reverb return on one track of a two track or cassette and your original dry signal on the other. Bring it back to your place and pan the dry signal to the center of your monitors, the reverb return from their plate on the right. Send the dry signal to your plate, return the reverb to your mixer, and pan it to the left. Now you can directly compare your plate to theirs, and tune and EQ until the sound of yours equals or betters theirs (subjectively). (A tape like this is available in the kit offer. It is made from an EMT 140 ST tube unit—the "classic plate" sound, and an Audi-ence solid state stainless steel unit. Notice the deep, smooth bass, and crisp, sparkling highs—as well as smooth decay. Sends chills up your spine, huh?) So use your ears and you can't go wrong.

A Case For The Plate

Theoretically, you are done—but you really need a place to put your unit and something to put it in. The best place would be a separate quiet room or closet so no outside vibrations or sounds will affect the plate—even though a case for the unit is recommended. The case is simply a wooden box that the frame can sit in—or better yet is suspended in. EMT, Ecoplate and Audi-ence use pressboard, Lawson uses plywood, while Stocktronics has only paneling. (See *Figure 7*.) Use whatever is fitting for you. The frame can be placed in the case on rubber "feet" or better yet suspended in the case using rubber shock absorber mounts on bolts placed through holes drilled in the frame or rubber straps with hooks as found in automotive stores for holding down luggage, etc. The straps can be wrapped around the frame and the hooks hooked to holes or eyelets in the case. This way you can literally pound the case with little vibration to the frame and plate. Eyelets can also be put on the outside of the case on each end so rods can be inserted for carrying. If you are only using the plate during mixdown (the usual

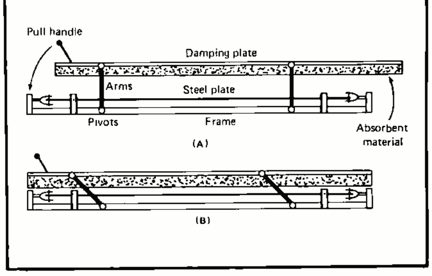

Figure 8. Drawing of damping system.

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situation), the studio isn't a bad place to leave the unit. It probably has the best isolation from your monitors and has easy access to your mic inputs and headphone jacks. The case only has to be a few inches bigger than the entire unit on each side unless you plan on using the next step—damping.

Damping

The decay time for the reverb as it stands is approximately five sec. at 500 Hz. This is fine for some applications, mostly jazz and classical, but can easily be varied by fitting a damping plate. This can be a piece of plywood, etc. the same size as the plate covered with an absorptive material; compressed fiberglass, styrofoam, foam rubber, etc. that can be moved closer or further from the plate to absorb the vibrations, and therefore, shorten the decay time of reverb. EMT, Lawson, Audi-ence and Ecoplate all move the damping plate in parallel to the steel plate from almost touching (⅛-inch) to 6 to 8-inches away. This is accomplished by forming a parallelogram type setup where two metal arms attach to the frame and to the damping plate so when the damping plate is moved the arms travel sideways and move it closer in parallel to the steel. (See *Figure 8.*) Stocktronics simply hinges their styrofoam damping plate at the bottom with a piano-type hinge and pull the top closer or further to the steel, claiming a more uniform frequency response in the decay characteristics. (See *Figure 9.*) A handle or lever attached to the damping plate facilitates moving the damper. It can also be remote controlled using servo motors and cams, but this is beyond the scope of this article. The choices of materials, method, or even use of damping at all is left up to you.

Plate-Tricks

Using EQ will help you get the reverb characteristics you are after much easier than tuning alone. In fact, all the commercial units have some sort of EQ in their electronics, either a bass cut off on the pickup amps or a high frequency boost to the drive signal or any and all combinations. It is better to add the high frequency boost to the drive, since the "hiss" will be lost in the plate, not picked up by the pickups, maintaining a good signal to noise ratio. Low cut can be done on the return as well as the drive signal. EMT cuts the bottom at 80 Hz, but many engineers use a 700 Hz high pass filter to accentuate the top. EMT uses a drastic drive EQ—as much as 20 dB boost between 10k and 15k and above. All plates of various brands have the curve they use to give their brand a specific sound. We found this to be the hardest area to explain. It is hard to EQ a plate to sound right if you are not familiar with the sound of one. Therefore we have developed a "passive drive signal response shaper" patched between the board send and the drive amp. It automatically makes the plate sound like a reverb instead of a piece of steel in a frame. The curve is fixed and works especially well when used with the driver and pickups we recommend. You just plug the output of the send to the input of the response shaper; the response shaper to the drive amp; the amp to the driver and then attach the pickups to the plate, return the pickup output to the board and you have reverb. If you have an EQ or two to spare, it would be a good idea to patch one to the send and one to each return. This will allow you to match the sound of almost any of the commercial plates or any plate sound you have heard. In fact, Studio Technologies, makers of Ecoplate, have released a product with a parametric EQ on the send and one on each return, as well as a delay and noise gate all in one unit. The delay and gate help with some plate "tricks" you can use for popular effects you are probably familiar with. "Delayed Send"—Take the send to the plate and first put it into your delay line. Use a full bandwidth setting so you don't lose any top end. The effect is that you're in a large hall where the first reflection isn't heard until milliseconds after the initial dry signal. The longer the delay, the bigger the hall. A good example of an extra long pre-delay is the reverb on the snare on the end of "It Keeps You Running" by the Doobie Brothers. You can hear the snare hit and the reverb later. Sort of "boom...cha!" This extra long pre-delay will also bring out the deficiencies of a reverb unit, and if you try it on a "twangy" spring, the time delay doesn't let the program mask the "boing" of the snare transient. But with a plate, this is no problem. A shorter pre-delay also gives the effect of a longer decay time because the decay will end 50 ms later if it starts 50 ms later.

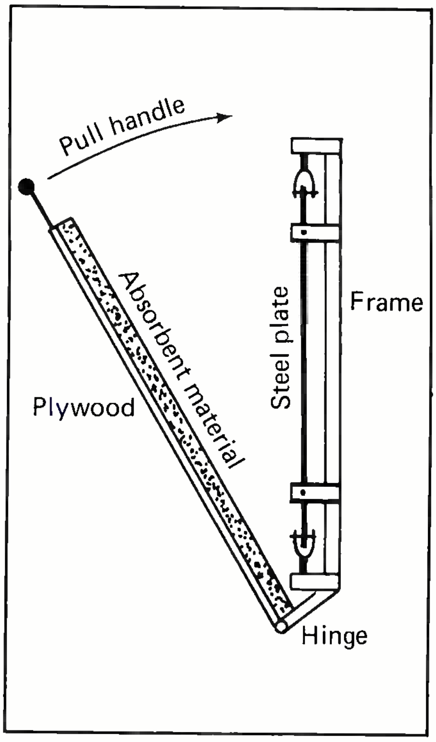

Figure 9. Drawing of piano hinge-type damping system.

To shorten the decay without damping, a noise gate comes in handy. Placed on the return, the release time can be shortened and when the attenuation and threshold are properly set, the decay will be gradual and smooth, only shorter. If the controls are set to dramatically attenuate the decay, the effect can be rhythmic. For example, if hand claps are done on the downbeat, the reverb decay can end sharply and completely on the upbeat—one and two and... clap-cha clap-cha. Mic-mix has recently released a gate designed specifically for this purpose.

You can also gate the send to the plate so you only reverberate certain signals. For example, if you want reverb only on the snare track, and the snare wasn't gated when recorded, if you gate the track to the send, you will only get the reverb on the snare beats, not on any tom toms, bass drum, or cymbals that might have leaked onto your snare mic.

Experiment and you can get any sound you've heard, and some you can invent.

...So if I just use one plate reverb with a lot of top EQ and a gate for the snare, another one with a lot of bottom for "thunder toms" and one with a long pre-send delay and high frequency boost for that "sizzily vocal" sound; maybe one for the strings...with maybe a little flange on the return...and maybe one more...

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Harry Langdon Photography

Steve Levine

Digital On Demand

Steve Levine is a producer that comes complete with a full array of digital recording equipment, free of charge. When Steve produces a project, it is almost always recorded digitally. This is illustrated by the unique clarity and impressive sound he produces along with his regular engineer, Gordon Miline. And the sound quality Steve achieves on his work is just as impressive as his credits. Among the work he has produced is both Culture Club albums, albums by David Grant, the Melody Makers (consisting of Rita and Bob Marley's children), Quarterflash and the Beach Boys, as well as his own solo LP. In the future, Steve would like to get into underscoring and movie soundtracks.

Steve credits Bruce Johnston of the Beach Boys for giving him the encouragement necessary to make the move from engineer to producer. Steve's first meeting

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with Bruce came when Bruce was at CBS Studios producing an album for a band called Sailor and Steve engineered some of the tracks. Bruce is also responsible for planting the first ideas of working with the Beach Boys into the producer's head. It was the first time the band had any contact with digital recording and Steve says it was an exciting prospect. It would follow that the Beach Boys—with their full and numerous backing harmonies—could use a producer like Steve Levine whose trademark is his lavish and well textured vocals.

Steve's approach is very technical. He attended an electronics technical school when he was younger and then started working at CBS Studios in London immediately after school. As Steve explains, "Although it wasn't 'formal training,' I learned a fair amount from working at CBS. I very quickly picked up the methods that were required." Steve worked his way up through the ranks. After working at CBS Studios as an engineer for a while, Steve felt he wanted to progress further, "So production was obviously the next avenue to take," Steve continues.

And obviously, Steve was right. He has been extremely busy over the last few months with the Beach Boys project and then working on a new album for the rock group Quarterflash at Miraval Studios in France. He is very much in demand—since he offers artists with a low budget the opportunity to record digitally—but he says he will choose his next project wisely and carefully. *MR&M* spoke to Steve during the Quarterflash project in France.

Modern Recording & Music: Do you work out of a home base studio?

Steve Levine: I worked continually from Red Bus Studios, but now I'm currently in the throws of building my own studio in London. The stage that it's at so far is that the walls are up. I've got a shell so far but it's being made as we speak.

MR&M: What kind of equipment do you plan on using?

SL: It's going to be 48-track digital with two Sony PCM-3324s with 3310 remote units, and two 1610s for mixing, which is my current setup at the moment because I have all of that equipment in flight cases and I take it to whichever studio I'm working in. I also have 5850 U-matic recorders and a DAE-1100 editor. I'll be building all that into the studio.

MR&M: How much engineering do you do?

SL: Well, I just generally oversee it and then I do the mixing.

MR&M: Why do you feel it's necessary to own your own equipment as opposed to either renting it or leasing it?

SL: Because by owning it, I can offer that service to the band, whereas a lot of bands are not in the position to use that equipment, because at the moment digital equipment is very, very expensive to use—so it sort of doubles the recording budget in the studio. And a lot of studios don't have the facilities to offer. So by owning it myself, there's a personal decision for me to record everything digitally because I believe that is the best method to go, and it's vastly superior to analog. I thought, 'Well if I have the equipment then I can offer that equipment as opposed to persuading the record company that that's the best avenue to take.' They don't have any problems because it's my equipment, so it's not as if they suddenly have to pay $300 an hour studio time or whatever a digital recording rate is. We can work in a normal studio, but have the most incredible results.

MR&M: How do you feel about CDs? Do you feel it's worth the extra money and work to have something produced for CD?

SL: Absolutely. Without question. And it also follows that it's worth spending the money on the recording process so that the CD can be the finest that you can possibly make because even from an analog master a CD is superior to a record. But if the original recording is completely digital, then the CD is absolutely perfect. When I listen to a CD that I've recorded digitally, it's like listening to the master. There's no difference which means that any person can go and buy the CD and enjoy what we enjoyed in the studio.

MR&M: When an artist first contacts you and asks you to produce them, what do you go about doing pre-production wise? Preparing yourself for the artist and such?

SL: Well, obviously I'd meet them and meet the various members of the band—if it was a band. It really stems from a lot of different avenues because obviously with a band like Quarterflash, they presented me with a set of demos, for my approval, and I was very impressed with the demos and then I took it a stage further, to meet the band and discuss how we would go about recording.

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And also they wanted very much to have a different sound which is why they came to me. But it could quite conceivably be that in the case of a solo artist there may not be the songs, it may be the record company, *X*, they want me to produce this artist, but they also want me to look for the songs and get the band together and all that sort of thing. So it really is no set hard and fast rule, it really depends on what the project is.

MR&M: When an artist approaches you, do they usually have something in mind, or do they prefer to leave the actual direction up to you?

SL: Normally, they leave it up to me because that was the whole point in coming to me, because obviously, they're familiar with things that I've done and perhaps they're familiar with some articles that I've worked on and they know the processes that I work with. In Quarterflash's case they were more than aware that I record digitally and I use a lot of digital techniques in recording and so that was the whole point in getting the situation together.

MR&M: Do you get the final say?

SL: Well, I think the final say is a very gray area. Final say should always be a group situation. There's no point in a producer handing a band their record on a plate because that's not the point of doing it. I mean the whole point of a producer is to bring the best out of the artist and not for the producer to present the artist with their finished record. So the

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final say is obviously a completely joint decision, which is one we're all happy with. But obviously the idea is for a producer to indicate to an artist certain areas that they may pursue that they may not have possibly considered.

MR&M: How do you handle an artist that is very difficult to work with?

SL: I haven't actually had one yet, so hopefully...I think you can tell from your initial meeting with an artist whether they're going to be difficult or not. I don't like working with people that are not hard workers. I think there's absolutely no point. Recording records should be a pleasure and not a pain in the ass and if an artist's a real pain in the ass, I'd rather not work with him. Because there's no point. I'm doing it because I enjoy doing it, so we might as well all have a really good time when we make the record and come out of it feeling that we've made the best record we can and we enjoyed the recording process as well. That's not to say we don't have discussions or *heated* discussions about the way things should be, I think that it's only positive that when you have a lot of creative people in the same room, there's bound to be tension and friction, but that's very positive as opposed to somebody who's an artist that's on an immense ego trip or something.

MR&M: Is there anyone you'd like to produce that you haven't yet?

SL: That's an open ended question, I don't actually know, really. There are a lot of people I'd like to work with and one thing that is excellent about being successful is you do have the opportunity to work with them. On the Beach Boys album, for example, I got the opportunity to work with Stevie Wonder, which was something that I always wanted to do and that was really excellent. So I think just being a producer does give you the opportunity to at least meet these people and work with them which is really good.

MR&M: How did working with the Beach Boys come about?

SL: Well, it was a slow laborious process to make it come about. It started mainly when Bruce Johnston planted the original seeds on doing it —he suggested it to me. Wouldn't it be a good idea,' and I thought, 'Yes, wouldn't it be a good idea.' But it took a long time for negotiations to get to the point that we were anywhere near getting into the studio. Mainly because they're so un-together [the Beach Boys] it just took a long time to physically get them all in the same place at the same time. But eventually it all came together. Initially it started more than a year ago—the earliest tracks I started were the ones I started with Carl—the backing tracks to "It's Getting Late," and "Where I Belong"—when he came over to London. Then Carl went back to America and Brian came over, and I did backing tracks for most of his songs that are on the album, and I then carried on working on the other tracks on my own, and then I eventually went to America with finished backing tracks and then we started to voice what was already done. Then we recorded new in LA, "Getcha Back," and "She Believes In Love Again." Those are the only two tracks we recorded in LA.

MR&M: What studios did you use?

SL: We used Red Bus and CBS Studios in London, and Westlake Audio in Los Angeles.

MR&M: How long did the recording take?

SL: Totally, it took about nine months. That wasn't on and off all the time. That was fairly large chunks. That was the longest project I ever worked on. The majority of the time was taken because they are notoriously slow and we took ages to get backing vocals. One thing that I wanted to achieve—the world remembers the Beach Boys for being a lot better than they actually are. When you actually analyze the Beach Boys, particularly vocals, they're really not that good, I mean they could sing, but their intonation, their tuning and just their basic need to work is very low down on their list of priorities. So you have a band that makes an awful lot of money going on the road playing old hits, with a standard that's normally pretty low. So obviously, why do they need to sit in a studio at *X* thousand pounds a day when they could be out on the road doing that. So that's the attitude of half of the band. Then you get the attitude of the other half of the band who wants to work, so that caused a lot of problems there. It was Bruce and Carl who wanted to work the most and of course I was accused of favoritism because I was spending more time with Bruce and Carl because they were prepared to be in the studio more than the others. And then we'd need Michael for a vocal and we couldn't find him for a couple of days or we'd need Brian and Brian wasn't around for a couple of days. It was all that sort of nonsense which eats up studio time at an alarming rate. Also, in the meantime, I got the call to go and see Stevie Wonder about his particular track so that was quite nice. That was actually a lot of light on the project, because it was very fulfilling working with him because he was such a fantastic person. Very often when you meet people that you've admired for years, they're often a great disappointment and you think, 'My God how did these people ever get on...why are they successful?' But he is one of the few people that I've met that was better than I'd imagined. He was so incredibly professional. It was so nice.

MR&M: Tell us a little about the actual recording.

SL: Well, that depended on each track. Musically, the majority of the tracks were done in London by myself and Julian Lindsey. We just recorded the tracks and when we needed specialist instrumentation—such as on "Maybe I Don't Know"—we wanted a rock type of guitar—and since I had worked with Gary Moore, and I really liked his particular style, I thought it would really suit that track. So I phoned him up and got him down to work on it. We managed to get Ringo Starr to play drums on "California Calling" because I met him in Washington last July 4th and he expressed an interest in working on the project. That was the ideal track for him to work on.

So, generally, I prepared the tracks in the studio to a fairly finished standard and then I brought in additional musicians where they were necessary, so I wasn't wasting their time and they could hear the track the way I intended it to be or

As an album on its own it stands up, but considering what we all went through, then I think it's fantastic.

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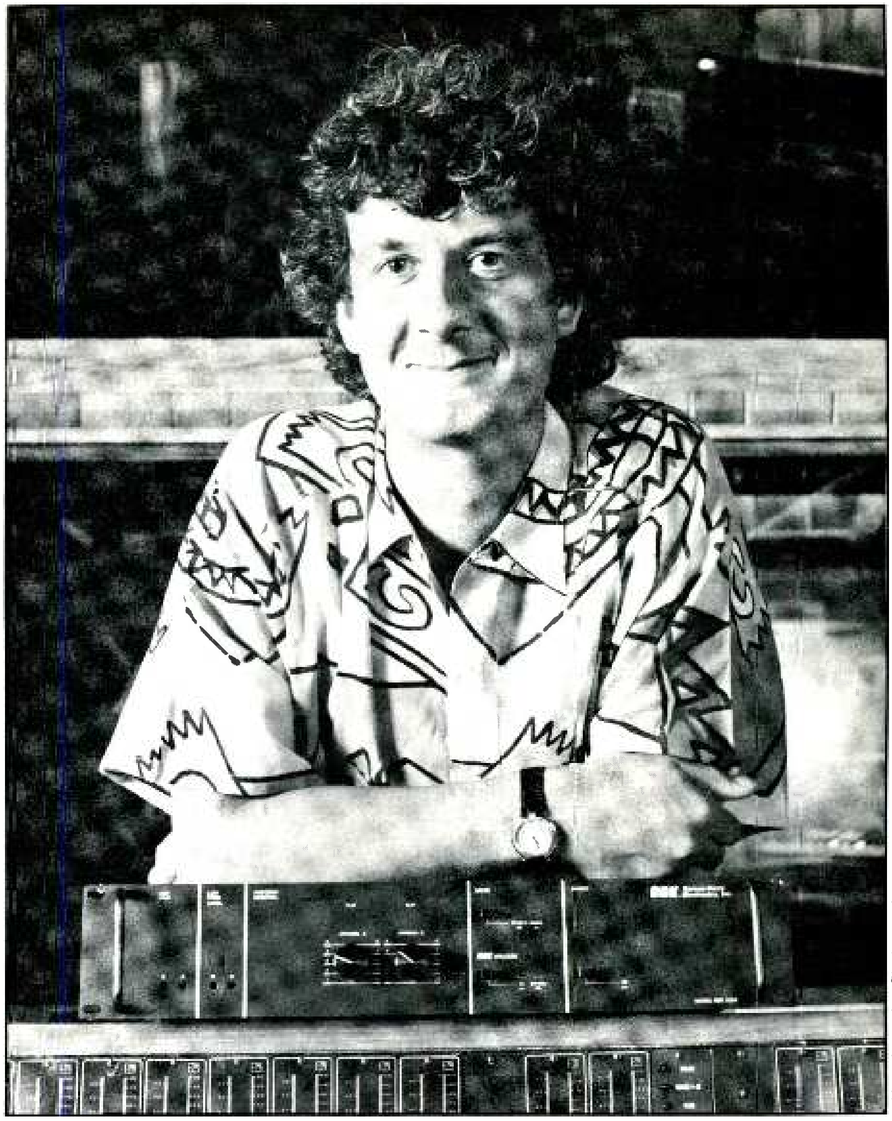
how the song was intended to be. And then they could put their performance on top of that. So in the case of say one of Brian's songs, he would show me what the basic part was for the keyboard and he would play one part...so we'd program one particular part and then we'd augment that bit by bit—either he'd play some additional things or we'd play some additional things. And the same is true with Carl—although a lot of Carl's songs were based on guitar so we translated the guitar information into keyboard information and then played accordingly. Bruce was no problem because he is a very good keyboard player so his was the easiest backing tracks to do because his understanding of music and keyboard playing was the best. Stevie Wonder was no problem since he played everything on that track himself. Roy's [Hay, from CC] song was no problem because he played most of the instruments on that track himself. That's how we went about doing the main backing tracks.

I put the vocals on in Los Angeles and that process depended on what we did. In most cases it was necessary for either Carl or Bruce to do their part of the backing vocal first mainly for tuning because I found it difficult for them all singing at the same time—if I could get them all physically in the same place at once was an achievement—but normally I couldn't so I found it was better to record Carl or Bruce first and then track on each person separately. That meant I was more in control of their vocal level and their harmony tuning. So I think overall the vocals sound fantastic. I mean they're really in tune and absolutely superbly well recorded and I think that was the benefit of doing them separately. There were one or two simple blocks where we did do them all at once because it was quicker and it was such a simple part and there was no problem with it. But generally as a rule we did them separately. And then we did lead vocals after we did all the backing vocals, and then I returned to London to bounce all the vocals down because, as you can imagine, the vocals were spread across a lot of tracks and they needed tidying up. And also there were subtleties missing and some of the concepts of some of the tracks had changed a little bit over the time. After I heard a track with all the vocals on it, I slightly changed my interpretation of the track a little bit and then I added a dobro when we got



Harry Langdon Photography

Steve Levine with the BBE unit which he uses quite often.

back to London. It seemed to be a joining link between the harmonica and the way the harmonies had turned out. There were just a few little bits and pieces that had to be added here and there.

Then we were ready to do the mixing. The mixing was done half in London and half in Los Angeles. That was due to a lot of things, mainly people being in the right place at the right time. And that's how we finished the album. But it was a very long and slow process—just very hard work.

At many points in the recording—particularly in LA—I sort of wondered if I got myself in a situation where we'd never, ever have this record out. Sometimes they'd be such a pain in the ass, I just didn't know what to do. There were times I felt like saying, 'I'm just gonna walk out and leave you 'cause I don't need this.' I'd flown all that way to make an album and they just couldn't be bothered to make an album. They'd be busy on the phone talking about other investments, and their horses or whatever.

When you work with a young band they can't wait to get into the studio and to record, but when you work with a band that's spent so long in studios they are sort of bored with the process, plus the process was so technical—with the use of digital recording and the use of the Fairlight—it was completely over their heads (with the exception of Carl and Bruce) and they didn't have a clue on what was going on.

I think that frightened them a lot to realize they had gotten so out of touch with recording. They were once leaders in recording so I think that they think that they will always remain so. When they realized on this project that they were actually light years away from what was happening, I think that really did frighten them a lot. So that probably created an additional fear.

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But the album did get finished and I'm very, very pleased with the results, and considering the circumstances under which it was done, it makes it even better than it is. As an album on its own it stands up, but considering what we all went through, then I think it's fantastic.

MR&M: What mics did you use?

SL: I used a Sanken, a Japanese microphone that's actually very good because it contains two capsules—one for the bass and one for the top end. The Neumann TLM-170 was particularly good. Those are the mics I used for virtually all the vocals. I am very keen on getting the mic to match the vocal before I then have to start to process the sound—if at all. So generally I just swap between the two and different people's voices do sound slightly different on the microphones. So I just put both of them up for each vocal and I just use the one that sounds the best. There was no general rule that I used.

MR&M: How did you record the drums?

SL: The set up was continually changing. On Ringo Starr's kit, for example, I wanted to get an old-fashioned type sound so I used two mics—an Electro-Voice RE-20 for the kick drum and for overheads we used a pair of Neumann 87s. On "Maybe I Don't Know," which was more of a rock'n'roll thing. I used a much larger selection of mics—along the lines of [AKG] 414s, the Sanken for some snares, and [AKG] 451s for high-hats. For the toms I got a very good drum sound with the Sonys, but it was very varied. One thing that worked very well was to use the Sony C-48 in figure-8 position in between the toms so that the mic was in the center and the figure 8 pattern was reaching the right hand and left-hand toms. I was quite pleased with the sound I got from that. On most of that I didn't even have to use any EQ.

> ***The video is only a marketing tool. I mean there have been instances where good videos have sold poor records, but ultimately, the most successful records have been the ones that were actually good records.***

MR&M: There's a very distinctive clarity on the vocals of the records you produce. Do you attribute this to the fact that you record all the vocals in stereo, or because of the digital recording?

SL: It's both of the effects. Firstly, it comes from a choice of a very, very good microphone, because the better the quality of the microphone, the better your starting point is. So that's why you must use a very good transparent microphone. Also, I don't ever-EQ things because a lot of people tend to really EQ vocals beyond all recognition and that only causes problems later. Secondly, obviously digital recording helped because you don't have any problems with bouncing because bouncing degenerates vocals excessively on analog equipment whereas digital eliminates that problem because there are no generations involved.

MR&M: How do you feel about the new age of MIDI, digital, and the like?

SL: I'm very pleased because at least it's a step in the right direction. I don't think it's the total answer, but it is a step in the right direction. I have experienced, though, incompatibility problems between machines and certain machines behave very slowly so you end up with excessive delay problems with some equipment. But the mere fact that you can at least plug one machine into another and get it to work, reasonably well, even if it doesn't work exactly, at least it's working better than it did two years ago when the thing wouldn't interface with something else. So it is a step in the right direction but I think we are in the very early stages at the moment and till every single machine has the same format as every other machine, we are still not completely interfaced, but we are closer. So it definitely is a step in the right direction.

MR&M: Do you ever produce with video in mind?

SL: Not really, because my role in life is to make records and I think videos are a separate field and they have their people that are particularly good at making videos. I mean if I was into making videos then I'd be a film director. But obviously when you're making a record, certain images do suggest themselves because you have an intimate knowledge of the track. You sometimes think, 'Oh, well that's a really good accent and it would be nice if, when using a video, a particular element of the record would be brought out in terms of the editing of the footage or something.' So obviously you tend to think of ideas as you go along. But as to what the final result is, obviously that's down to whoever is directing the band or who the record company chooses.

MR&M: So you feel video is secondary?

SL: Oh, absolutely. The video is only a marketing tool. I mean there have been instances where good videos have sold poor records, but ultimately, the most successful records have been the ones that were actually good records. I think particularly in England, there seems to be a slight problem budgeting videos. It seems to be perfectly acceptable for a band to spend $100,000 making one single video whereas if you spend that amount of money making one single record, the record company will go mad. So I think priorities seem to be in the wrong place at the moment. You get particular record companies who would say, 'We can't afford to record digitally it's ridiculous' and yet they can afford to spend all that money on a video. So I think there seems to be a slight imbalance—I don't know whether that's the case in America—I know videos in America are made for a lot less money than they are in England, but in England I know that's the case. Bands seem to be spending an excessive amount of money on videos, but not so much on their records. They seem to have their priorities in the wrong place. They're very excited about making a video and probably less excited about making a record. They can't wait to be, sort of film actors or something. I can't think the two marry themselves together particularly well. You can't be good at everything so either you're a very good artist or a very good actor. And I've yet to see anybody who's good at both.

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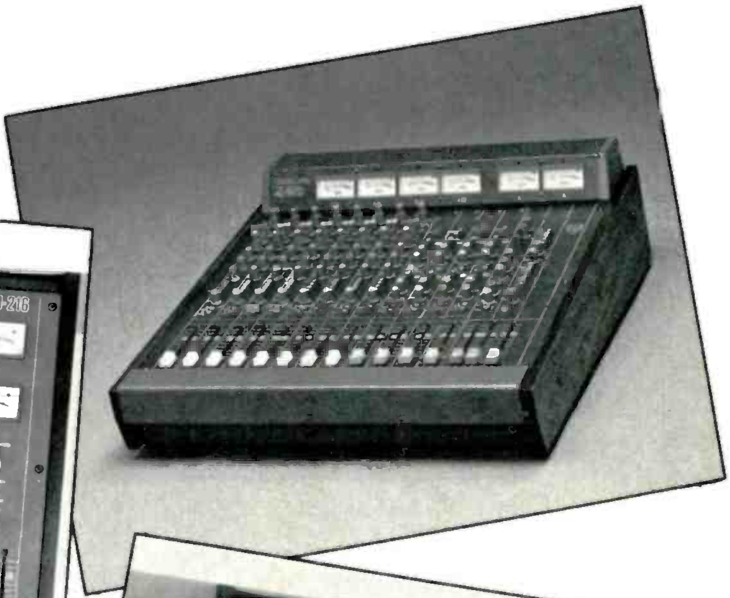
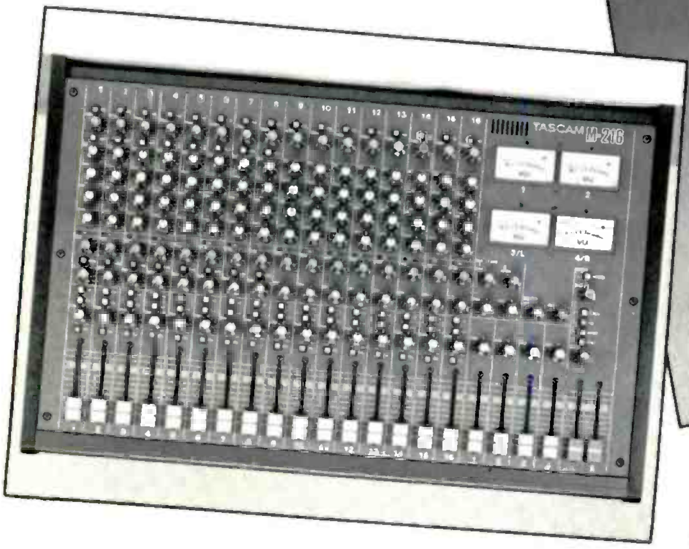
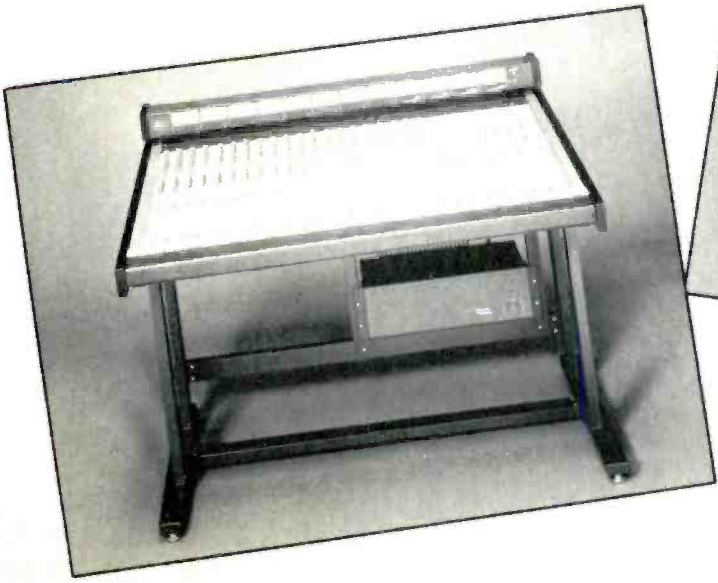
MODERN
RECORDING
& MUSIC

Looks At

Consoles







& Mixers

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All dimensions are W X H X D in inches

ALLEN AND HEATH BRENELL

SR SERIES

This is designed for sound reinforcement and 2 or 4-track recording. All boards feature 4-band EQ, 4 auxiliary sends, 100mm slide faders, 48V phantom power, stereo and mono outputs. It is available as the SR-8, 12, 16 and 24-input configurations. The SR-416 and SR-424 are available with 4 additional submaster channels, EQ bypass and muting switches on all input channels.
PRICE: SR-8—$1,300; SR-12—$1,600; SR-16—$1,950; SR-24—$2,700; SR-416—$2,700; SR-424—$3,500.

SYSTEM 8 SERIES

These are designed for use in 8 and 16-track recording as well as sound reinforcement. Input configurations of 16 and 24-inputs are available and may be expanded by linking System 8 mixers together or using the EX-8 expander module, all without sacrificing any inputs or outputs.
PRICES: SYSTEM 8-168D—$4,500; SYSTEM 8-1616D—$5,100; SYSTEM 8-2416D—$6,300; SYSTEM 8-EX8—$1,950.

OMC SERIES

These are microprocessor controlled input/output designed mixing boards for 8, 16, and 24-track recording. They feature 3-band sweepable EQ, 6 aux sends, 8-, 16-, 24-channel automated buss assignment and muting. May be synced to tape using external computer and AHB software package.
PRICE: OMC-16—$4,300; OMC-24-$5,800; OMC-31—$7,600.

SRM SERIES

On stage monitor mixers in an 18 X 6 or 24 X 8 configuration. Includes a passive microphone splitter system, engineering monitor section, external power supply and road case.
PRICE: SRM-186—$3,600; SRM-248—$4,800.

SYNCON-B SERIES

Syncon Series B is available in 2 frame sizes, which can be supplied in any combination initially or retrofitted on site at a later date. All formats are wired for 24-track routing. P&G faders standard.
PRICES: B 12-8/8—$9,980; B 24-16/16—$16,860; B 36-24/24—$24,700; B 36-32/24—$30,120; B 48-44/24—$39,980.

ARIA

AMX 160B

This is a 16 X 2 mixing console with spring reverb, 3-band EQ, 5-band stereo master EQ, and sum output. It also has 16 balanced inputs, 2 XLR outs, 16¼-in. phone line ins, 16¼-in. phone channel outs, 2¼-in. phone master meters (left, right, monitor), 2 RCA outs and headphone jack.
PRICE: $949.

AMX 120B

This is 12-channel version of the AMX160B above.
PRICE: $749.

AMX 80B

This is an 8-channel version of the AMX160B above.
PRICE: $649.

AUDIO CENTRON

100 SERIES

This series features 8, 12, and 16-channel stereo mixers in X 2 X 1 configurations. Each channel has low and high-impedance inputs plus an insert jack for channel effect patching, a variable trim control with LED indicator, 3-band EQ, reverb/effect and monitor sends, pan control, and a 60mm fader. The master section has assignable 9-band graphic EQs, plus 10 stage LED output meters.
PRICES: $899 to $1,299.

STAGE/STUDIO SERIES

These mixers come in 16 X 24 X 4 X 2 X 1 which can also be used in 16 X 24 X 6 X 1 for additional submix capability. Each channel has assignable low and high-impedance inputs, send and return jacks, 3-band quasi-parametric EQ, 2 pre and 2 post sends and 100mm faders.
PRICES: AC416—$2,995.95; AC424—$3,995.95.

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BIAMP SYSTEMS

83B SERIES

These are live oriented (rack & console) 2 submaster mixers that feature discrete bipolar balanced inputs (with patch), floating/balanced outputs, 3-band EQ, 100mm faders and one monitor and one effects buss (drives internal reverb routed to subs and monitor). Noise/distortion equivalent to studio consoles (EIN = -129 dB, IMD = .01%). The dimensions are: 683B (rack) 6-inputs and 2 X 1 outputs; 19 x 10½ x 4¼; 883B (rack) 8-inputs and 2 X 1 output; 19 x 10½ x 4¼; 883B (console) 8-inputs and 2 X 1 outputs; 23¼ x 4½ x 19; 1283B (console) 12-inputs and 2 X 1 outputs; 29 x 4½ x 19; 1683B (console) has 16-inputs and 2 X 1 outputs; 34½ x 4½ x 19; 2483B (console) has 24-inputs and 2 X 1 outputs; 45¾ x 4½ x 19.
PRICES: 683B—$799; 883B rack—$999; 883B console—$1,199; 1283B—$1,499; 2483B—$2,199.

24 SERIES

These are submaster consoles for live and recording and feature discrete bipolar balanced inputs (mic/line with patch, direct out and tape-return), floating/balanced outputs, 3-band EQ (sweepable mid), stereo monitor (source pre/post & tape-in), 2 effects (source pre/post), phantom power, direct to L/R channel assignment and solo. Noise/distortion equivalent to studio consoles (EIN = -128 dB, IMD = 0.01%). The dimensions are: 824 has 8-inputs and 4 X 2 & 1 outputs; 24¼ x 5½ x 25; 1224 has 12-inputs 4 X 2 & 1 outputs; 30 x 5½ x 25; 1624 has 16-inputs and 4 X 2 & 1; 35½ x 5½ x 25; 2424 has 24-inputs and 4 X 2 & 1 outputs; 45¾ x 5½ x 25; 3224 has 32-inputs and 4 X 2 & 1; 57 x 5½ x 25.
PRICE: 824—$1,499; 1224—$2,249; 1624—$2,999; 2424—$4,249; 3224—$6,199.

28 SERIES

This group is identical to the 24 Series except with 8 submasters and no direct to L/R channel assignment. Both feature 14 segment dual color fluorescent displays, 100mm faders, aux returns for each submaster and extensive switch selectable signal routing. The dimensions are: 1228 has 12-inputs and 8 X 2 & 1 outputs; 35½ x 5½ x 25; 1628 has 16-inputs and 8 X 2 & 1 outputs; 41 x 5½ x 25; 2428 has 24-inputs and 8 X 2 & 1 outputs; 51¼ x 5½ x 25; 3228 has 32-inputs and 8 X 2 & 1 outputs; 62½ x 5½ x 25.
PRICES: 1228—$2,779; 1628—$3,359; 2428—$4,599; 3228—$6,699.

BIMIX SERIES

These are modular 16 buss in-line, I/O (input/output) recording consoles in 3 frame sizes. Features include 3-band sweepable EQ, 2 effects and 2 cue busses (switchable), meters for all inputs/outputs, solo and mute. The dimensions are: 840 has 8-inputs and 4 X 2 X 1 outputs; 23½ x 8¼ x 29¼; 1280 has 12-inputs and 8 X 2 X 1 outputs; 23½ x 8¼ x 29¼; 1612 has 16-inputs and 12 X 2 X 1 outputs; 35½ x 8¼ x 29¼; 2016 has 20-inputs and 16 X 2 X 1 outputs; 35½ x 8¼ x 29¼; 2416 has 24-inputs and 16 X 2 X 1 outputs; 53.5W x 8.25H x 25¼; 2420 has 24-inputs and 20 X 2 X 1 outputs; 2816 has 28 inputs 16 X 2 X 1 outputs; 2824 has 28 inputs and 24 X 2 X 1 outputs; 3216 has 32-inputs and 16 X 2 X 1 outputs; 3224 has 32-inputs and 24 X 2 X 1 outputs.
PRICE: 840—$3,999; 1280—$5,299; 1612—$6,499; 2016—$7,699; 2416—$9,999; 2420—$9,999; 2816—$11,049; 2824—$11,049; 3216—$12,099; 3224—$12,099.

CARVIN

MX-1688

This recording console has 16-channels with 8 outputs directly compatable with any 8-channel recorder. Features include 3-band parametric EQ on each channel, independent 8 into 2 monitor mixer, 4 busses and studio feed with source selection. The dimensions are 35⅛ x 8¾ x 29.
PRICE: $2,995.

MX-1644

This sound reinforcement console has 16 channels with 4 sub-outputs mixing to stereo with mono capability. Features include 4-band EQ on each channel, 4 busses, 2 monitor and 2 effects, channel and subgroup patching, and stackable inputs. Frequency response is 15 Hz - 25 kHz +/- 1 dB. The dimensions are 35⅛ x 8¾ x 29.
PRICE: $1,995.

MX-2422

This sound reinforcement console has 24 channels by stereo output/feeding a mono outputs with balanced inputs and outputs. 2 onboard graphic EQs, high quality dynamic VU meters, 3-band EQ on each channel with sweepable mid-range, soloing, 2 monitor sends, 1 effects send, built-in reverb are also included. The dimensions are 39 x 8½ x 25.
PRICE: $1,999.

MX-1622P SOUND REINFORCEMENT CONSOLE

This sound reinforcement console has 16 channels, stereo output/feeding a mono output. Features built-in talkback mic, individual channel patching, built-in power amplifier with 400 watts RMS power, mic/line switching and built-in reverb system (Hammond). The dimensions are 29 x 8½ x 25.
PRICE: $1,599.

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MX-1222P

This sound reinforcement console is a 12 X 2 X 1 format console. It features built-in power (powered models only) 400 watts stereo power. 3-band EQ w/sweepable midrange, 2 monitor sends, 1 effects send with built-in reverb. The dimensions are 29 x 8½ x 25.

PRICE: $1,299.

MX-822P

This sound reinforcement console has an 8 X 2 X 1 format system. It features 400 watts stereo power, XLR inputs and outputs, 3-band EQ with sweepable mid-range, mic/line switching, phantom power and channel patching. The dimensions are 23 x 8½ x 25.

PRICE: $949.

MX-622P

This sound reinforcement console has a 6 X 2 X 1 format with built-in stereo graphic EQ, XLR inputs and outputs, 2 monitor sends, 1 effects send, with built-in reverb. Built-in phantom power, headphone output and VU meters. The dimensions are 23 x 8½ x 25.

PRICE: $799.

CP-600

This sound reinforcement console has 6 input channels, monaural output, powered with 150 watts RMS. Features include graphic EQ for output, 2-band tone controls, built-in reverb with 1 effects send and XLR inputs and outputs. The dimensions are 24 x 7 x 11½.

PRICE: $399.

CREST

TEN SERIES

This series is designed primarily as a small front of house board or as a keyboard/drum submixer. It is available in 8, 12, 16, 24 and 8-channel rack mount versions. Features include mic/line switch, 4-band EQ, 3 aux sends, internal reverb, solo system and summed mono output.

PRICE: $1,200-$2,800.

FIFTEEN SERIES

This series is a 12-channel rack mount board designed for high quality submixing or as a stand alone rack mixer. Features include mic/line switch, 30 dB pad, 4-band EQ, 3 aux sends, internal reverb solo system, insert points, and 48V phantom balanced outputs.

PRICE: $1,600.

DEAN MARKLEY

SPECTRA PM400A

This is a 4-channel sound reinforcement mixer with 85 watts RMS at 4 ohms. Each channel contains 2 high-impedance inputs, input level, low and high EQ with 15 dB boost and cut plus reverb/effects send. The 5-band graphic EQ is centered at 100, 330, 1k, 3k and 10 kHz with a boost and cut of 12 dB. The dimensions are 15⅞ x 4½ x 13⅜.

PRICE: $389.

SPECTRA PM600A

This is a 6-channel sound reinforcement mixer with 160 watts RMS at 4 ohms. Each channel contains 1 high-impedance and 1 transformerless balanced low-impedance input, input level, monitor send, low and high EQ with 15 dB boost and cut and effects send. The 9-band graphic EQ is centered at 63, 125, 250, 500, 1k, 2k, 4k, 8k and 16 kHz with 12 dB boost and cut. The dimensions are 21½ x 5⅛ x 15¾.

PRICE: $649.

SPECTRA PM800A

This is an 8-channel sound reinforcement mixer with 160 watts RMS at 4 ohms. Each channel contains one transformerless low-impedance linput, one high-impedance input, input level, monitor send, low, mid and high EQ with 15 dB boost and cut and effects send. The dimensions are 25¼ x 4⅜ x 17½.

PRICE: $795.

DOD

R-855 MIXER

This is a four-input, stereo output unit with effects loop. It provides up to 48 dB of gain. Auxiliary inputs allow the mixing of a signal from a tape deck or other source, or for bridging two or more R-855s for more inputs.

PRICE: $269.95

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R-855 XLR
This unit offers the same features as above, but features all XLR inputs and outputs for permanent installations and professional recording use.
PRICE: $299.95

ELECTRO-VOICE
8400 SERIES MIXERS
All models have left, right, mono, monitor, aux 1 and aux 2 outputs, all with unbalanced and transformer isolated connections. Features include 48V phantom power, 3-band EQ with switchable midrange center frequency, send and return jacks on each input and pre-fader solo system.
PRICES: 8408 (8-channel)—$3,210; 8416 (16-channel)—$4,185; 8424 (24-channel)—$6,160; 8424 (32-channel)—$7,700.

8200 SERIES MIXERS
Same features as the 8400 Series, above.
PRICES: 8208 (8-channel)—$2,140; 8212 (12-channel)—$2,500; 8216 (16-channel)—$3,165.

8108
This is an 8-input, 4-output rack mountable mixer for use in sound reinforcement, recording, broadcast or zoned systems. All inputs and outputs are balanced and transformer isolated. Other features are LED bar graph PPMs and a 2-band shelving EQ on each channel.
PRICE: $1,275.

FENDER
42 SERIES
This is a series of multi-purpose mixing consoles which feature low noise (-130 EIN), low distortion and high output capability (+ 24 dBm), allowing these consoles to be used in a wide variety of applications. The 4216 is a 16-channel unit, the 4212 is a 12-channel unit and the 4208 is an 8-channel unit.
PRICE: 4216—$1,699; 4212—$1,099; 4208—$899.

FOSTEX
2050
Compact, line level mixer, 8 X 2, with independent program and cue mixes. Two or three units may be patched in parallel or in series for versatile grouping. A stereo input source can be added and mixed to the 8 line inputs via front panel jacks and controls. The dimensions are 14 x 1¾ x 6½.
PRICE: $995.

MN-50
This is a 5-input mono line mixer. It can be powered with AC adapter or 9V battery. There are 4 line inputs with mic or line input with selectable input pad. Also features an internal compressor with variable release, and on or off The dimensions are 1¾ x 6¾ x 4.
PRICE: $55.

450 MIXER
The 450 mixer is an 8 X 8 X 4 X 2 board with application for any type of mixing. It has 8 low-impedance balanced transformerless mic inputs with 48V phantom power, individually selectable. The 450 has 8 channels of 3-band sweepable equalization, 4 panable effects returns, channel in & out solos, 3 effects sends, overload lights, and onboard headphone amplifier. The dimensions are 21½ x 21¾ x 4½.
RPICE: $995.

GALAXY AUDIO
M802
This is an 8-channel, console or rack mount mixing board with balanced mic & line inputs, 3-band EQ, peak detector lights on each channel, two sub groups output and 48V phantom power. The weight is 25 lbs. and the dimensions are rack: 19 x 15 x 4; console: 18½ x 16 x 9¼.
PRICE: $775.

M1202
This is a 12-channel version of the M802, above. The weight is 35 lbs. and the dimensions are 25½ x 19 x 5¼.
PRICE: $999.

M1602
This is the same as the M1201 above, but it is a 16-channel version with a weight of 37 lbs. and dimensions of 31¼ x 19 x 5¼.
PRICE: $1,450.

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HOLMES

XM-1200PM

This is a 12-channel mixing console, expandable with identical features via Holmes XM-100 expansion module. Features include dual 135 watt power amps (powered monitor section), dual 7-band graphic EQ, dual 10-step LED VU meters and high and low-impedance inputs. This is also available in a 12-channel, single 250 watt power amp model.
PRICE: XM-1200PM—$1,449.95; Single 250 watt—$1,249.95.

XM-600PM

This is a 6-channel version of the XM-200PM, above.
PRICE: $949.95.

XM-900PM

This is a 9-channel mixing console, expandable with identical features via Holmes XM-100 expansion module. Features include dual 135 watt power amps (powered monitor section), dual 7-band graphic EQ, dual 10-step LED VU meters and high and low-impedance inputs.
PRICE: $999.95.

TECH 8000

This is an 8-channel mixing console, with single 135 watt power amp, 5-band graphic EQ, 5-step LED VU meter, high and low-impedance inputs, patching network, trim controls and 2-band channel EQ.
PRICE: $699.95.

NEI

MODELS 244XM AND 164XM

Both units feature transformer balanced inputs and external power supply with 48V phantom power. Input channels feature high, low (2 position shelving) and sweep mid EQ plus effect and monitor sends with aux send switchable pre or post. The units are 24 X 4 X 2 X 1 and 16 X 4 X 2 X 1 respectively. The 244 XM is 47½ x 28 x 6½ and weighs 90 lbs. The 164XM is 35½ x 28 x 6½ and weighs 70 lbs.
PRICE: 164XM—$3,695; 244XM—$5,195.

MODELS 1622, 1222, & 822

All three units feature transformerless balanced inputs and optional external power supply with 48V phantom power. Input channels feature 3-way EQ plus effect and monitor sends with aux send switchable pre or post. The units are 16 X 2 X 1, 12 X 2 X 1 and 8 X 2 respectively. The 1622 is 33½ x 23½ x 5 and weighs 40 lbs. The 1222 is 27½ x 23½ x 5 and weights 35 lbs. The 822 is 20 x 23½ x 5 and is 21 lbs.
PRICE: 1622—$1,825; 1222—$1,525; 822—$1,025.

MODELS 1623, 1223

Both units feature transformerless balanced inputs and outputs plus internal phantom power. Input channels features 3-way EQ plus effect, monitor and reverb send. Units are 16 X 2 X 1 and 12 X 2 X 1. Outputs are via Neutrix NC3MP connectors and are balanced for sub 1, sub 2, monitor and main. The 1623 is 27¼ x 23½ x 5 and weigh 35 lbs.; the 1223 is 23 x 23½ x 5 and is 30 lbs.
PRICE: 1623—$1,495; 1223—$1,195.

MODEL SR60

This unit features transformerless balanced inputs. Input channels feature 3-way EQ plus reverb/effect and monitor send. Master section has 10-band, octave EQ. The dimensions are: 13¼ x 19 x 12 and the weight is 37 lbs.
PRICE: $695.

MODELS SR61, SR62

These powered mixers feature transformerless balanced inputs. Input channels feature 3-way EQ plus reverb/effect and monitor send. Master section has 1-band, octave EQ. Available with one (SR61) 100 watt RMS amplifier or two (SR62) 100 watt RMS amplifier (mono-bridgable). The dimensions are rack mount: 19 x 12 x 12; box: 13¼ x 19 x 12. The weight for the SR61 is 43 lbs. and the SR62 is 44 lbs.
PRICE: SR61—$795; SR62—$995.

MODELS 611P, 881P

These powered mixers feature transformerless balanced inputs. Input channels feature 2-way EQ plus reverb/effect and monitor send. Master section has 10-band, octave EQ. Available with one (611P) 100 watt RMS amplifier or two (811) 100 watt RMS amplifier (mono-bridgable). The dimensions on the 611P are 17¼ x 19½ x 5 and weights 27 lbs.; the 811P is 20 x 19½ x 5 and weighs 34 lbs.
PRICE: 611P—$775; 811P)$1,075.

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PANASONIC

WR-8616

This is a post production/recording console featuring 16 electronically balanced inputs, 4 group outputs, left and right stereo outputs, 48V phantom power at each input, 2 stereo switchable pre/post sends, choice of mono or stereo inputs, and choice of basic or tape monitor group modules. The dimensions are 35¾ x 10½ x 29¾ and the weight is 114 lbs.
PRICE: $4,125.

WR-T812 and WR-T820

These are 8 group output recording consoles with 12 inputs tailored for 8-track recording (WR-T812), or 20 inputs for 8- or 16 track recording (WR-T820). Both feature extensive metering capabilities and ability to simultaneously mix incoming signals with tape playback during overdubbing without patching, 3-band sweepable EQ, 48V phantom power and stereo solo on all input channels. The WR-T812 is 31 x 6 x 31⅞ and weighs 70 lbs.; the WR-T820 is 42⅛ x 6 x 31⅞ and weighs 70 lbs.; the WR-T820 is 42⅛ x 6 x 31⅞ and is 99 lbs.
PRICES: WR-T812—$4,040; WR-T820—$5,060.

WR-8210A

This is a recording console with 10 inputs, 4 group outputs, 48V phantom power on all input channels, left right outputs, 10 tape recorder sub mix inputs to monitor tape or echo returns, LED metering direct outs and insertion points on all inputs. The dimensions are 24¾ x 5¾ x 20¾ and weighs 36 lbs.
PRICE: $1,995.

WR-8112 and WR-8118 SOUND REINFORCEMENT/RECORDING CONSOLES

These sound reinforcement/recording consoles feature 18-channel inputs, 4 group, 2 master and 1 mono master output (WR-8118) or 12-channel inputs, 4 group, 2 master and 1 mono master output (WR-8112). Features include 48V phantom power, 3-band sweepable midrange EQ, push button tape input (allows multitrack monitoring for overdubbing and mixdown), 3 sends and solo on all channels. The dimensions of the WR-8112 are 24¼ x 5¾ x 20¾ and is 37 lbs.; the WR-8118 is 35⅜ x 5¾ x 20¾ and weighs 58 lbs.
PRICES: WR-8112—$2,500; WR-8118—$3,150.

WR-S208, WR-S212 and WR-S216

These stereo mixing consoles feature 2 channels on each mixer with stereo line and phono inputs. Each balanced mic & line input features 48V phantom power, 3-band sweepable mid range EQ, 3-send circuits for effects and monitoring and LED metering on all outputs. The dimensions are as follows: WR-S208: 8 inputs X 2 X 1 output; 17 x 5⅞ x 20¾ and weighs 33 lbs.; WR-S212: 12 inputs X 2 X 1 output; 25⅝ x 5⅞ x 20¾ and weighs 40 lbs.; WR-S216: 16 inputs X 2 X 1 outputs; 30⅝ x 5⅞ x 20¾ and weighs 47 lbs.
PRICE: WR-S208—$1,295; WR-S212—$1,795; WR-S216—$2,095.

WR-130

This audio mixer has 8 inputs and 2 outputs. Features include mic/line or phone inputs, 2-band EQ, pre or post send control, peak LED on all input channels and VU metering on main outputs. Suitable for remote broadcasts, recording or sound reinforcement. The dimensions are 18¼ x 6 x 16⅛ and weighs 18½ lbs.
PRICE: $995.

PEAVEY

MD SERIES

There are 8-, 12- and 16-channel versions. Each channel features one high-impedance and one low-impedance input, pre EQ send and return jack (¼-inch stereo), input sensitivity/gain, monitor send, low and high active shelving EQ controls, peak/notch type mid frequency control, effects level control, 1 stereo pan control, calibrated channel slider (level) control. The dimensions are: MD-8 is 20⅞ x 4⅝ x 22½ and weighs 26 lbs.; MD-12 is 26⅞ x 4⅝ x 22½ and weighs 30 lbs.; MD-16 is 32⅞ x 4⅝ x 22½ and weighs 37 lbs.
PRICE: MD-8—$639.50; MD-12—$799.50; MD-16—$949.50.

MARK III SERIES

There are 12-, 16- and 24-channel versions. Each channel features one high-impedance, one low-impedance transformer-balanced input, pre and post send and return, input attenuation, 4-band EQ, 2 pre monitor sends, 2 post effects sends, pan control, slider level controls, LED indicator. MARK III-12 is 29⅝ x 6¾ x 29 and weighs 67 lbs.; MARK III-16 is 35⅝ x 6¾ x 29 and weighs 77 lbs.; MARK III-24 is 47⅝ x 6¾ x 29 and weighs 97 lbs.
PRICE: MARK III-12—$1,499.50; MARK III-16—$1,799.50; MARK III-24—$2,399.50.

MARK IV SERIES

There are 16- and 24-channel versions. Each channel features one high-impedance input, one low-impedance transformer-balanced input, pre and post send and return, 4-band EQ, 2 pre monitor sends, post effects send, pan control, push button channel assignment to 4 submasters, direct sum assignment and LED status indicator. The dimensions are: MARK IV 16 X 8 is 29⅛ x 7 x 43¼ and weighs 87 lbs.; MARK IV 24 X 8 is 29⅛ x 7 x 54¹⁄₁₆ and weighs 111 lbs.
PRICE: MARK IV-16—$2,499; MARK IV-24—$2,999.

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XR-700

This is a dual powered system with power amps for main and monitor. Each power amp produces 100 watts RMs at 0.1% THD into 4 ohms. 7 channels, monaural, effects level, high, mid, low active EQ, monitor send, and output attenuation on each channel. Master section features effects, return, reverb return, effects reverb to monitor, reverb contour, and main, monitor and effects master levels. The dimensions are 24¼ x 7¼ x 22⅛ and it weighs 46 lbs.
PRICE: $749.

XR-800

This unit has 100 watts RMS per channel at 0.1% THD into 4 ohms, DDT compression, 8 channels, stereo, effects master, pre fader listen level, headphone level and jack and master slide attenuator. There is also a 9-band graphic EQ for channel A and channel B. The dimensions are 24¼ x 8⅜ x 29⅝ and weighs 60 lbs.
PRICE: $1,099.50.

XR-1200

This unit has 200 watts RMS per channel at 0.1% THD into 4 ohms, DDT compression, 12 channels stereo, input attenuation, monitor A&B send, high, low and mid active EQ, effects level, pre fader listen button and slide level control on each channel. Master section features monitor, an effects send, level, pan, 9-band graphic equalizer for channel A and channel B. The dimension are 28¾ x 8⅜ x 29⅝ and weighs 84 lbs.
PRICE: $1,399.50.

MARK IV MONITOR MIXERS

These are 16- and 24-channel versions. Each channel consists of the following: bridging male and female XLR low-impedance transformer-balanced inputs with ground isolation, one high-impedance unbalanced patch, pre send and return, input gain/attenuation, 4-band EQ, mute, phase reverse, and pre fader listen push buttons, 8 monitor matrix level controls and LED status indicators.
PRICE: MARK IV-16 X 8—$2,499; MARK IV-24 X 8—$2,999.

MD MONITOR MIXERS

These are 12- and 16-channel versions. Each channel consists of the following: bridging male and female XLR, low-impedance balanced inputs with ground isolation, pre send and return, input gain/attenuation, 3-band EQ and 6 monitor matrix level controls. Each of 6 master sections contain one low-impedance unbalanced output, LED ladder display and variable low cut filter. The dimensions are : MD monitor 12 X 6 is 30⅞ x 4⅝ x 2½ and weighs 34 lbs.; MD monitor 16 X 6 is 36⅞ x 4⅝ x 22½ and weighs 38 lbs.
PRICE: MD 12 X 6—$799.50; MD 16 X 6—$949.50.

PULSAR

SERIES 80 TOURING CONSOLE

The series 80 console is primarily a live, touring console that can also be used for 8-track recording. You can get up to 8 independent mixes of the 8 sub groups simultaneously and independently of one another. Features include 5-band sweep or graphic EQ, balanced line and mic in, 16 balanced outputs, access in/outs on all modules, phase switch, 4 aux sends with EQ, high pass filter, cue, mute, full headphone monitoring and overload metering on all gain stages.
PRICE: 20-frame—$9,000; 28-frame—$11,200; 32-frame—$12,400.

RANE

MM 12 MATRIX MIXER

This is a rack mountable console providing 8 X 6 matrix control over 12 input channels (each with mic split, gain, 3-way EQ, cue, loop and aux) and 6 output channels with cue, expand, EQ loop and 2-stage parametric EQ for each. The dimensions are 19 x 21 x 2½.
PRICE: $1,399.

ROLAND

PA-250C

This sound reinforcement console is an 8-channel powered mixing console rated at 250 watts (125 x 2 at 4 ohms). It features 8 balanced and unbalanced inputs, a stereo 9-band EQ, phone inputs, recording outputs, stacking jacks, and high and low active EQ on each channel.
PRICE: $1,295.

PA-150C

This sound reinforcement console is an 8-channel powered mixer featuring the same control configurations as the PA-250C and has a power rating of 150 watts RMS, (75 watts x 2 at 4 ohms).
PRICE: $1,095.

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CPM-120
This sound reinforcement console is a compact 8 X 2 stereo powered mixer (60 watts x 2 at 8 ohms). Each channel features input attenuation, high and low EQ, effect level, panpot and level controls. The output section features a 7-segment LED array, left and right output controls, and a headphone jack/level control.
PRICE: $675.

BOSS BX-800
This is an 8-channel stereo mixer that features 8 high-impedance inputs, 2 high-impedance outputs and effect send/return jacks. Each channel has input gain, high and low EQ, effect level, panpot and slider level controls. It weighs under 5 lbs.
PRICE: $360.

BOSS BX-600
This is a compact 6-channel stereo instrument/line mixer that features 6 high-impedance inputs, 2 high-impedance outputs and peak LED indicator. Each channel has input gain, effect level, pan and level controls. It also has stereo effect returns.
PRICE: $225.

BOSS BX-400 MIXING BOARD
This mixer is useful in multiple keyboard set-ups. It is a 4-channel mono mixer with each channel featuring a 3 position input switch (mic/inst/line) and a level control. It weighs less than 2 lbs.
PRICE: $130.

BOSS KM-04
This is a 4 X 1 instrument mixer for use in multiple instruments set-ups. The battery powered unit is also useful as a pre-amp for amplifying acoustic guitar or bass pickups.
PRICE: $60.

SECK

SECK-62
This portable mixing console has 6 balanced mic/line inputs into 2 outputs with insert point on each input and 3-band EQ with sweep mid range. There are 2 pre-fader sends and 2 post-fader sends, stereo inplace solo on all inputs, auxiliary sends, and returns.
PRICE: $1,345.

SECK-122
This is the same as the SECK-62, but with 12 inputs.
PRICE: $1,995.

SECK-1882
This has 18 balanced mic line input to 8 subgroups to 2 outputs, 48V phantom power, full in-line monitor section on all 18 inputs, 3-band EQ with sweep mid range, stereo reverb and foldback on monitor section, and one foldback and 2 auxiliary sends on each input. An optional meter bridge is available.
PRICE: $3,995.

SHURE

M67
This is a remote mixer specifically designed for use in remote broadcasting, studio recording, sound reinforcement, and as an add-on mixer for expanding existing facilities. It features a frequency response (+/- 2 dB from 30 to 20,000 Hz) with low distortion, up to + 18 dBm output, and low noise and RF susceptibility. Four low-impedance balanced microphone inputs (one convertible to line level) are provided. Each input has a switchable low-cut filter.k The dimensions are 11⅜ x 2¾ x 7½ and weighs 4 lbs. 13 oz.
PRICE: $400.

M68
This unit can be used to provide additional microphone inputs to another mixer. Each of four microphone level inputs has its own switch for selection of low-impedance (balanced or unbalanced) or high-impedance (unbalanced) microphones, and a high level auxiliary input. It has two outputs: one is a microphone level output (low or high-impedance), the other is a high-impedance, high-level output. The dimensions are 11⅜ x 2¾ x 5¼ and weighs 4 lbs.
PRICE: $205.

M67
This compact, professional, four input microphone mixer features mix bus, simplex (phantom) power, automatic muting circuit, active gain controls, electronic power supply regulation, and a dramatic reduction in distortion. Both high- and low-impedance microphones can be used at the same time. The dimensions are 12¼ x 3 x 9 and weighs 4 lbs, 2 oz.
PRICE: $257.

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SOUNDCRAFT

SERIES 200B

This series is available in 8, 16, and 24-channel mainframes. (8 channel available in rack mount version.) Fully modular with 4 group outputs and stereo master module. Standard features include 8-track monitoring facility, 4-band EQ, VU metering, 4 aux sends (switchable), and internal headroom level of + 26 dB.

PRICE: 8 X 4—$2,500; 8 X 4 rack mount—$2,500; 16 X 4—$3,750; 24 X 4—$5,250.

SERIES 400B

This series is fully modular and offers 4-band EQ, high pass filter, gain control, 4 aux sends (two switchable), 4 outputs, stereo master module. Also available in monitor version. Effects return module also available as an option.

PRICE: 16 X 4 X 2—$5,750; 24 X 4 X 2—$7,750; 32 X 4 X 2—$9,950.

SERIES 500

This is a fully modular sound reinforcement console available in 16, 24 and 32-input mainframe sizes. Eight subgroups with 8-track monitoring (switchable effects returns or tape returns) and 2 aux sends per group, VU metering, switchable pre or post fader or pre or post EQ aux sends (6) on each input channel.

PRICE: 16 X 8 X 2—$7,250; 24 X 8 X 2—$8,950; 32 X 8 X 2—$11,500.

SERIES 600

This is a fully modular recording console available in 16, 24 or 32-input mainframe sizes. Eight subgroups with 16-track monitoring and individual EQ, aux sends (2), pan and soloing facility. Switchable pre or post fader or pre or post EQ aux (6) sends on each input channel. LED metering, 4-band EQ, 48V phantom power.

PRICE: 16 X 8 X 2—$7,950; 24 X 8 X 2—$9,950; 32 X 8 X 2—$12,500.

SERIES 800B

This fully modular sound reinforcement console is available in 16, 24, 32 and 40-input mainframes. Eight subgroups with direct outputs for 8-track recording. Matrix module available allowing for 8 discrete mixes. Also available in monitor version.

PRICE: 16 X 8—$12,950; 24 X 8—$16,500; 32 X 8—$22,500; 40 X 8—$25,950.

SERIES 1600

A fully modular recording console available in 16, 24, and 32-input mainframes with 8 outputs and 16-track monitoring or 24-track monitoring with patch bay. Includes stereo and aux busses.

PRICE: 16 X 8—$12,950; 24 X 8—$16,500; 24 X 8 (with patch bay)—$19,950; 32 X 8—$22,500; 32 X 8 (with patch bay)—$25,000.

SERIES 2400

A fully modular recording console for 24-track recording, available in 24 and 28-input mainframes. Extensive monitoring facilities allows monitor channels to be used as effects returns, giving up to 56 equalized inputs on the 28-input version. Available with both LED and VU metering.

PRICE: 24 X 16 X 24 VU (24-track monitoring)—$29,500; 28 X 24 VU—$39,500; 28 X 24 LED—$43,500.

TS 24

A fully modular in-line recording console available in 24, 32, 40, 48 and 56-input mainframe sizes. It features a master status switch which reconfigures the console with one button. Also available as a television production console, designated the TV 24. Automation available.

PRICE: (non automated consoles): 24-input—$46,500; 32-input—$53,500; 40-input—$62,500; 48-input—$75,000; 56-input—$80,000.

TASCAM

SERIES 200, M-208

This is designed for live mixing, sound reinforcement with additional production and recording applications. It has 8-input, 4-buss output, effects buss, separate foldback mixer, solo, balanced O dBV inputs and outputs, optional rack mount and 3-band EQ. The dimensions are 17¼ x 5⅛ x 16½ and the weight is 18¾ lbs.

PRICE: $995.

SERIES 200, M-216

This is a compact 16-channel mixer for live PA applications or multitrack recording. It has 16-input, 4 buss output, effects buss, separate foldback mixer, solo, balanced O dBV inputs and outputs, and 3-band EQ. The dimensions are 25½ x 5⅛ x 16½ and the weight is 26½ lbs.

PRICE: $1,495.

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M106
This mixer is suitable for small scale production, PA or multitrack recording. Also very popular in nightclubs for location sound system applications. It has 6-input, 4 output buss, auxiliary and effects busses, 2 switchable meters and 2-bank EQ. The dimensions are 15¾ x 5⅓ x 15¾ and the weight is 15¼ lbs.
PRICE: $595.

300 SERIES, M-308
This is a dual purpose recording and sound reinforcement console with 8-input, 4-main buss outputs, additional stereo buss, full metering, 3-band EQ, pre-fader listen (PFL) solo, 2 aux busses, effects buss, after-fader listen (AFL) solo, 8-track monitor mixer, balanced input and output. The dimensions are 23 x 8¾ x 27¼ and the weight is 48⅛ lbs.
PRICE: $1,649.

300 SERIES, M-312
This console has 12-input, 4 program buss outs, additional stereo buss, 3-band EQ, pre-fader and after-fader listen, 4 aux busses, 1 effects buss, fully distributed monitor, balanced input and output, summed output buss and talkback facilities. The dimensions are 28¼ x 8¾ x 27¼ and the weight is 57⅛ lbs.
PRICE: $2,595.

300 SERIES, M-320
This is a 20-input version of the M-312, above. The dimensions are 39 x 8¾ x 27¼ and the weight is 79⅛ lbs.
PRICE: $3,495.

500 SERIES, M-520
This board is suited for studio recording, large scale production, or any use requiring flexibility. It has 20 inputs, 8 program buss outputs, 4 selectable aux systems, 3-band sweep EQ, stereo solo-in-place, fully balanced in and out, talkback, full 16-track monitor, and test tone oscillator. The dimensions are 43 x 9½ x 31½ and the weight is 104 lbs.
PRICE: $5,495.

500 SERIES, M-512
This is a 12-input, 8-track monitor version of the M-520, above. The dimensions are 31½ x 9½ x 31½ and the weight is 84 lbs.
PRICE: $3,995.

WASHBURN

BFE 6250
THis is a 6-channel dual powered board with 3 inputs per channel (balanced low-impedance, unbalanced high-impedance, and high-impedance in/out), 3-band EQ per channel, and a 10-band stereo graphic EQ that can be assigned to mains, monitor or any outboard source. Also includes onboard reverb with footswitchable effects and 150 watts per channel at 4 ohms.
PRICE: $899.

BFE 8250
Same as the 6250 above, but it is an 8-channel version.
PRICE: $999.

BFE 1240
Same as the 6250 above, but includes phantom power and 200 watts per channel at 4 ohms.
PRICE: $1,599.

BFE 8400
Same as the 1240 above, but it is an 8-channel version.
PRICE: $1,399.

BFE 1690
This is a 16-channel, unpowered board with a 3-band graphic EQ for pre or post shaping, 3 inputs per channel, individual cue switch monitors on each channel, and individual pan control on each channel.
PRICE: $1,599.

BFE 1290
Same as the 1690 above, but it is a 12-channel version.
PRICE: $1,299.

YAMAHA

MC CONSOLES
These have 12, 16 and 24-input channels with 4-group outputs and 2 stereo outputs. Electronically balanced, each channel has gain control with a 20 dB pad, 3-band EQ, 2 foldback and 2 echo sends, cue switch, modular construction, headphone and talkback section. There are patch points on each input channel and phantom power.
PRICE: MC1204—$2,095; MC1604—$2,695; MC2404—$3,795.

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MC MONITOR CONSOLES

These are 16 and 24-input monitor consoles with up to 10 independent mixes/outputs. Each input channel has electronically balanced XLR input and high-impedance input, 3-band EQ and 8 channel and 2 aux sends. Each of the 8 out masters has master on, cue and 80 Hz high-pass filter switches.
PRICE: MC1608M—$2,895; MC2408M—$3,995.

EMX POWERED MIXERS

The EMX 150 is a 6-input mixer with 150 watts per channel output. The EMX 200 (8 input channels) and EMX 300 (12 input channels) have 250 watts per channel output. There are 2 foldback sends, one echo send with built in analog delay, two 9-band EQs for output.
PRICE: EMX 150—$1,195; EMX 200—$1,595; EMX 300—$1,995.

RM RECORDING MIXERS

These are 16- and 24-input recording mixers for 8 and 16-track recording. Each channel has an electronically balanced XLR input and an unbalanced tape input, 3-band semi-parametric EQ, two echo sends and extensive signal routing. There is a built in patch bay on the console face.
PRICE: RM1608—$6,600; RM2408—$9,900.

PM2000

These have 24 and 32-input channels, 8 output and 8 matrix outputs, 12 VU meters, talkback and cue systems. Each input channel has 4-band EQ, 2 echo sends, 2 foldback sends and 100mm fader.
PRICE: PM2000-24—$29,500; PM2000-32—$34,500.

M1500 SERIES

This mixing console series is for fixed or portable sound reinforcement and production. It has 16, 24, and 32-input channels, 13 mixing busses, and a 4 X 4 matrix. Each input channel includes 4-band EQ, 2 echo, 2 foldback with 2 sets of inputs, talkback circuit, phantom power, and balanced inputs and outputs.
PRICE: M1515A-$11,000; M1524—$16,000; M1532—$19,800.

M916

This unit has 16 input channels, 11 mixing busses and a 4 X 4 matrix. Each input channel has a pair of selectable XLR inputs and patch points, 3-band EQ, 2 echo, 2 foldback sends, talkback circuit, transformer isolated inputs and outputs, and phantom power.
PRICE: $5,700.

M508/M512

These have 8 and 12 input channels with 4 mixing busses for fixed and portable sound reinforcement and production. Each input channel has one balanced XLR input, 3-band EQ, 1 echo, 1 foldback send and detented input level control and phantom power.
PRICE: M508—$1,650; M512—$2,300.

Allen & Heath Brenell USA Ltd.
5 Connair Rd.
Orange, CT 06477

Aria Music USA Inc.
1201 John Reed Ct.
Industry, CA 91745

Audio Centron
1400 Ferguson Ave.
St. Louis, MO 63133

Biamp Systems Inc.
PO Box 728
Beaverton, OR 97075

Carvin Mfg. Corp.
1155 Industrial Ave.
Escondido, CA 92025

Connectronics Corp./Seck
652 Glenbrook Rd.
Stamford, CT 06906

Crest Audio
150 Florence Ave.
PO Box 129
Hawthorne, NJ 07506

Dean Markley Inc.
3350 Scott Blvd. #45
Santa Clara, CA 95051

DOD Electronics
5639 So. Ripley Lane
Salt Lake City, UT 84107

Electro-Voice
600 Cecil St.
Buchanan, MI 49107

Fender Musical Instruments
1300 E. Valencia Dr.
Fullerton, CA 92634

Galaxy Audio
625 East Pawnee
Wichita, KS 67211

Holmes
3000 Marcus Ave. Suite ZW7
Lake Success, NY 11042

NEI
934 25th Ave.
Portland, OR 97232

Panasonic Industrial Co.
1 Panasonic Way
Secaucus, NJ 07094

Rane
6510-216th SW Suite D
Mountlake Terrace, WA 98043

RolandCorp US
7200 Dominion Circle
Los Angeles, CA 90040

Shure Bros.
222 Hartrey Ave.
Evanston, IL 60204

Soundcraft Electronics Inc.
1517 20th St.
Santa Monica, CA 90404

Tascam
7733 Telegraph Rd.
Montebello, CA 90640

Washburn International
230 Lexington Dr.
Buffalo Grove, IL 60090

Yamaha International Corp.
Pro Products Division
PO Box 6600
Buena Park, CA 90622

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bob buontempo

Poor Recorder's Almanac

Monitoring and Ambience

plug plug plug plug plug plug

Before we get down to business, let me get down to business. I am happy to announce that my partner in crime, er, musical endeavors, Les Miller and I just got signed to a recording contract with Atlantic Records.

We are signed as artists, and the name of the band we use is "Suburban Nites." Hopefully, by the time you read this column, you will have heard the record, especially if you are a "Honeymooners" fan. The title of the single is (The Honeymooners Theme)—"You're My Greatest Love" b/w "Madison Ave."

It is comprised of the updated theme music of the TV series with two actors doing their imitations of "Ralph Kramden" and "Ed Norton" quoting lines they made famous from the show.

It was recorded and mixed entirely at my studio, HomeGrown Studios, a 16-track facility.

All authorization from Jackie Gleason was obtained, and we just knocked on doors in NYC until we got signed.

Anyway, the main point of this is not so much boasting, (although of course we feel proud), but the proof of the saying with which I started out these columns: YES YOU CAN!!!

By the way, if anyone has a copy, (of course, if you don't, you can run right out and get one), and would like to know any specifics about a sound, instrument, piece of equipment, or how a sound was obtained, please feel free to write me here at *MR&M* for the info. Seriously, if you do get a copy, the long version (12-in. mix) has better sound quality than the 7-in. short version, (common to all analog recordings). Besides, if you buy one of mine, I'll buy one of yours.

This column was purposely picked to be one that could be a little short, while still completely covering the subject at hand, so I can get one weekend of vacation this summer, (although by the time you read this, the summer, along with my record if you don't buy it, will be a distant memory [Ed.—What record?]. (See how good I am at guilt; I should have been a mother, although I'm sure some people already have that opinion of me).

Anyway, can we get this damn thing started!!??

"Okay, turn up the U-87 room mics please." Ahh!! John Bonham lives again!!

"How about the four distant mics on the guitar amp?" "GREAT!!!" "Is this Olympic Studios, London or

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what!!??" "And why didn't you tell me Phil (Collins) and Peter (Gabriel) were on the premises!!"

This is exactly what happens in your cellar or bedroom studio when you try to use room mic'ing in a similar situation, right? Especially if the room is square or rectangular, and is very live and full of standing waves. (A good way to hear what standing waves sound like is to find a long, narrow stairwell and clap in it. The resulting 'Booiinngg' after your clap will be the standing wave. Of course if the stairway was designed with good acoustics in mind, the reverberation may sound great, and instead of using this as an example of a standing wave, it can be used as a great reverb chamber. (It is still rumored that Simon and Garfunkel used the elevator shafts at Columbia Studios for the reverb on the snare in "Bridge Over Troubled Water" and "The Boxer." In any case, your studio room, unless it was designed for great acoustics, will probably sound like crap, and when you go to add a distant mic, say on the drums, it will only be trashy, standing wave-ish, and unusably muddy.

On the other hand, if you have a great sounding room, by all means use it for distant mic'ing and room sound.

But for those of you who don't live in a Cathedral, my suggestion as to how to treat your room may sound dated, but it is as contemporary as you can get. Deaden it. All, that is, except for a small section with non-parallel walls that can be kept 'live' or 'echo-y.' You can use this for recording drums, percussion, horns, or vocals, if the situation calls for it.

And don't forget, if that stairwell or your bathroom sounds good, by all means, use it!!

But generally speaking, your bedroom probably doesn't have the world's best acoustics.

Besides, and this is where I throw in that "as contemporary as you can get" stuff; except for vocals, how many things do you record direct, especially if you own a drum machine and a Rockman!! Lester and I just completed a soundtrack for an Aerobics video cassette; and DIDN'T USE ONE MIC IN THE ENTIRE RECORDING!!

We used a drum machine, a synthesizer, direct bass, and direct guitar processed through a Rockman. All the ambience was either printed on separate tracks, (using electronic delays or plate reverb), recorded on the instrument as it went down, or added during the mix.

And let me tell you, they are some of the absolutely cleanest tracks possible. No leakage, and they could be brought anywhere to be remixed because the only difference would be in the other studio's machine, console, effects added to the mix, and the monitor, because the direct, unprocessed signal is unaffected by any room distortions, characteristics, or deficiencies.

Speaking of which, brings me to the other part of this column that I wish to discuss: Monitoring.

Take everything I just said and apply it to monitoring. Room resonances, standing waves, mud and muck, and how many people are in the control room at any given time all affect the sound coming out of the monitors, not to mention the monitors themselves.

A few years ago there was a big movement towards room tuning; which again makes sense in an airport hanger sized control room. But if your studio is the bedroom, then the bedroom closet must be your control room, and I'm not sure how much room you have for huge bass traps in the closet.

At the same time, there was a movement towards "Near Field Monitoring," which I jumped on right away, not because I thought there was nothing wrong with room tuning (although, how many people do you know that have perfectly tuned spaces in their living rooms or cars, and have the speakers positioned EXACTLY between the perfect spot to sit in to obtain a true stereo image). But I didn't have a lot of room for speakers to be hung and placed, and a lot of money for the EQs and servicing for them to be perfectly tuned. Besides, before I even knew anything about this stuff, having the monitors close to me made more sense.

They could be evenly spaced, LOUD, and act as giant headphones. So my recommendation is to place a set of "mid-size" monitors at ear level on top of (or thereabouts) your console/board/mixer. Don't put them TOO close, you want to hear SOME bass, but stay away from hanging them from the ceiling thirty feet away and EQing the piss out of them. Make your control room "living room dead," in other words, approximate the acoustics of the environment that the recordings will be listened to in, and try and recreate that space. There are some trademarked designs that take into account the sound reflections off of the console, and the echo of the back wall, such as the LEDE™ (Live End Dead End) design, but unless you can spend more for the treatment of the room than you can for that new digital reverb you've been creaming for, don't make this a top priority; as good as they are, they are also as expensive.

Just think of the monitors as reproducing the sound you want, independent of the room. Whatever you hear from the monitors is all there is; everything else passes by and is absorbed by the room, never again making it to your ears, and is unaffected or distorted by the room acoustics.

Apparently, many manufacturers are realizing that this is a viable market and have made just such monitors readily available, and at a fairly reasonable price. Trying to mention as many as possible, the list includes: the infamous Yamaha NS-10s (be sure to put tissue on the tweeters if you want to be 'hip'), Ramsa, Fostex, Teac, Tannoy (expensive, but worth it), JBL, Auratones (don't forget their nasty SC-5's, still a world standard reference 'car speaker' monitor, and INCREDIBLY inexpensive), E-Vs, TOAs, Little Reds by Audiotechniques (if you can still find them), and the list goes on and on.

Sorry if I missed anyone, if you have a favorite that you like, be sure to let me know. Braun and Radio Shack just popped into mind. In any case, the main thing is to make sure you like the way they sound in whatever you use as your control room, and finally, like the way they sound playing back your final mix in a variety of different people's houses, and the way your mix translates to other people's systems, speakers, cars, club systems, etc.

Next time, we will start to get heavy. Getting drum sounds, gating reverb, etc.

We are looking into the soundsheets so you can hear this stuff, but we still need more readers to request them. In the meantime there's always my record...

It's schooltime, so be prepared for some heavy work. And by the way, thanks to all for the support and kind letters. I really appreciate them, and will try to give all you loyal readers all I can and all you want to know until you ALL have a record deal. Which we WILL ALL GET, OKAY!!??

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The Market Place

what's new in sound and music

SOUNDCRAFT CONSOLE

The Soundcraft Series 200B is a small frame console for use in recording, video and post-production, broadcast, and, by utilizing the eight monitor returns for effects returns (allowing for eight extra line inputs), it can be used as a sound reinforcement console. Features include an internal working level of -6 dB (allowing a +26 dB internal headroom level), four-band EQ, VU metering and four aux sends, balanced tape returns, line inputs and mix output, channel prefade insert point, -10 or +4 internal switching, and switchable 48 volt phantom power. It also features the ability to direct assign to subgroups and stereo busses and 100 mm ultra-smooth action faders. The equalizer section provides optimum HF and LF shelving with high and low-mid peaking equalizers at 60 Hz, 250 Hz, 5 kHz and 12 kHz. The Series 200B is available in four different configurations. The 8/4 8-channel console retails for $2,500. It is also available in a rack mount unit for the same price. The 16/4 (16-channel) and 24/4 (24-channel) retail for $3,750 and $5,250, respectively.

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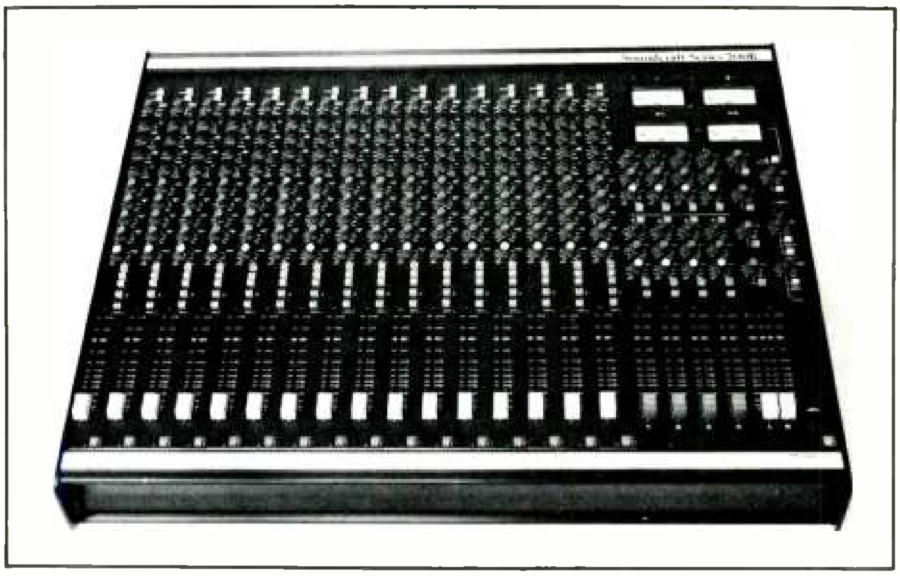
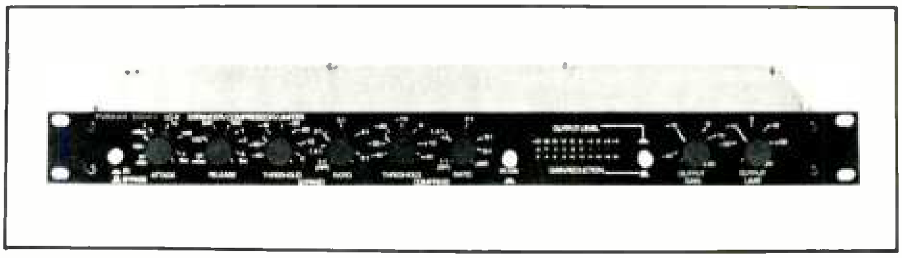


FURMAN SOUND EXPANDER LIMITER COMPRESSOR

Furman Sound's LC-X Expander-Compressor-Limiter is a multi-purpose unit that combines three sections which provide a total of six functions. These are: a variable ratio expander/gate, a variable ratio compressor/limiter/de-esser, and a fast responding peak limiter. The expander and compressor sections feature "soft-knee" action for natural, smooth sounding transitions. The peak limiter, designed to prevent the signal from rising above a calibrated threshold while preserving low distortion, features a "hard-knee" characteristic. Each section features its own threshold control and an LED which indicates the onset of each effect. In addition, the expander/gate and compressor/limiter/de-esser sections share a set of attack and release controls, making these sections either peak-or average-responding. For special effects, rear panel side-chain connections are provided which allow the unit to be used as a ducker for voice-overs, for frequency selective limiting, and for similar creative applications. All six effects are implemented by means of a single, high performance VCA which results in the unit's excellent performance specifications. This VCA, which is at the heart of the LC-X is an advanced design, discrete component, voltage controlled Attenuator (VCA). Using a technique Thermal Null System, developed by Furman Sound expressly for the LC-X, it automatically and continually corrects for voltage offset, minimizing distortion and other unwanted effects. This is responsible for the superlative transient response specifications of the LC-X. Other important standard features include: an output gain control (- to + 20 dB) for system level matching; a ten-segment LED meter, switchable to display either output level or gain reduction; two inputs and two outputs, (one instrument level and one line level); output muting to suppress turn on and turn off transients; a by-pass switch for instant A-B comparisons; a ground lift switch; and an interconnect jack for strapping two units together for stereo. Balanced inputs and outputs, and power supplies wired for 220 V 50 Hz are available as options. The LC-X retails for $449.



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THE PEAVEY CELEBRITY™ SERIES

Designed to have everything... except competition.

At Peavey Electronics we're dedicated to our commitment to design and manufacture high performance products at realistic prices. We've underlined that philosophy with our Celebrity Series line of microphones.

The Celebrity Series feature large diameter diaphragm/voice coil structures for increased sensitivity with the ability to handle high sound pressure levels. These higher output levels allow for significantly less mixer gain and are a tremendous aid in maintaining good signal-to-noise ratios.

Perhaps the most important characteristic of any performing microphone is reliability. The design of our cartridge/shock mount system increases ruggedness as well as isolation capability to insure long-term performance under severe field conditions.

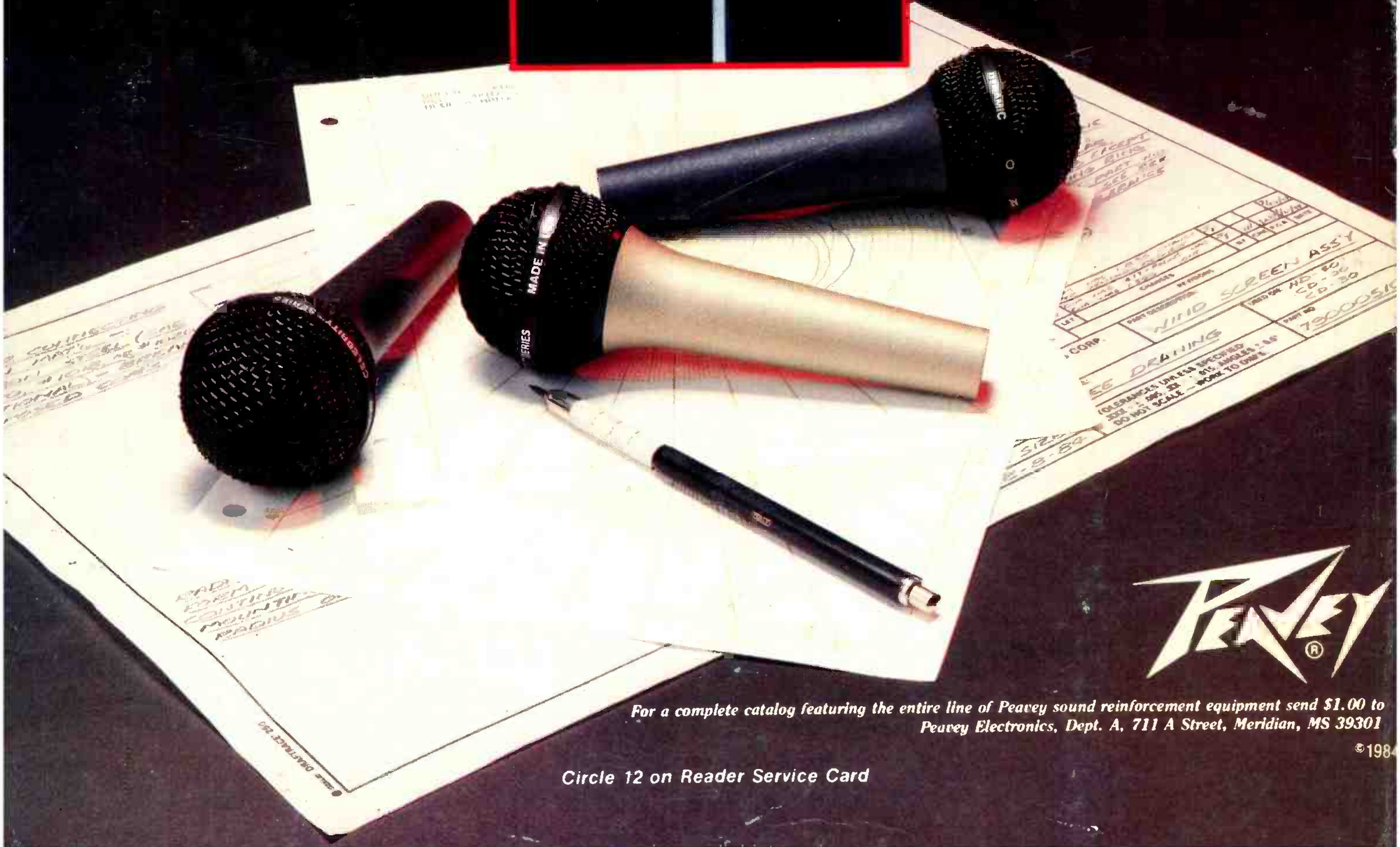
Our microphone screen utilizes extremely heavy gauge wire that has been "junction locked". Once the screen is formed, we do not stop there. The heavy wire screen is "fired" in an oven after forming, thus causing the plated wire to "fuse" at all interconnecting points. The result is an unbelievably durable "brazed" wire windscreen that will hold together under the most severe abuse. After the ball windscreen is formed, brazed and coated, a precision urethane foam pop filter is fitted to minimize the undesirable proximity effects. This special acoustically transparent foam protects the entire sound system by breaking up explosive high SPL pressure waves created by close vocals or close miking

percussion instruments. For those applications requiring even more acoustic screen from wind noise, etc., Peavey offers special external colored wind noise filters that slip over the screen and internal pop filter.

While outwardly, the appearance of the Celebrity Series is somewhat conventional, the aspect of "feel" has been given heavy emphasis since our experience has shown that performers prefer a unit that not only sounds right and looks right, but must also have a comfortable balance, weight, and overall tactile characteristics.

Special "humbucking" coils (models CD-30™ & HD-40™) have been designed into the microphone element that effectively counter-balance any hum that might be picked up from external sources. Performers who play clubs where hum from light dimmer switches or other sources are a problem can appreciate this unique feature.

We invite comparison of our Celebrity Series with other cardioid microphones. You'll see why we feel that in terms of performance, features, and price, there is no competition.



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