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STUDER REVOX

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JULY/AUGUST 1986 VOLUME 20 NO.4

the recording engineer

the sound contracting engineer

2 to 8 the smaller recording studio

ABOUT THE COVER

- Featured on the cover this month, Tiki Studios master control. See the feature article in this issue.

ABOUT THE 2-8 TRK COVER

- Featured on the 2 to 8 trk cover this month is sporting events composer/recordist Stephen Cullo's recording studio.

Two to eight trk cover photo by Michael Marzelli.

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Barry Blesser

Digital Audio

A New Way Of Thinking

ADOPTING NEW PARENTS

● One of the favorite subjects of writers in popular magazines and books is that of the future and changes. It is a topic that offers an opportunity to speculate in a provocative manner without being limited to any formal approach to the subject. Digital audio has certainly been a good example of change, and it has generated a large amount of speculation about what will become of our industry. In this month's column, I, too, will offer my insights.

The first distinction to be made is the difference between changes which come from new content and those which come from a new mode of existence. The first type of change requires the learning of new data, the latter requires a new way of thinking. Clearly, one is more difficult than the other. The invention of noise reduction equipment which could be added to professional tape recorders to improve the signal-to-noise ratio did not produce any new way of thinking. The method of recording was essentially the same, only the result was of a somewhat higher quality. The audio engineer did have to learn new content. The noise reduction equipment had certain artifacts which had to be understood. Competetive brands had different properties and some were better for certain applications. The introduction of this class of equipment did result in a change and the mastering of that change did require some efforts. Electronic digital editing, in contrast, requires more than a mastering of new content. It will result in a new way of thinking.

BIRTH PARENTS OF ENGINEERS

Audio engineering is a subset of the more general category of engineering. The broader profession has a long history. If we examine that history we discover that all engineering is the outgrowth of physics! A physicist is someone who is trying to master an understanding of the physical world. The content of the physical world is purely physical: electro-magnetic radiation, gases, liquids, solids, planets, chemicals, etc. The dramatic success

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and gain in these understandings over the last three centuries suggested that the insights could be used to improve the physical world of mankind. Engineers were born to use the understandings to manipulate the physical world to produce specific changes in the physical well being of mankind. Engineering was the application of this knowledge.

In the early history of engineering, most of the information was represented in the form of experience learned through an apprenticeship and through tables of data in handbooks. A high intellectual level was not required and the profession evolved from the supervision of the trade workers. This is a different professional history than the classical doctors, lawyers, and accountants. In the early twentieth century, Harvard had the opportunity to merge with MIT but refused, because they considered engineering to be a trade activity without intellectual substance. Engineerng required one to get one's hands dirty.

This was a good description up until the end of the second world war. By the 1950s it was clear that engineering had developed into a more powerful profession. This transformation was a change which came about through a new way of thinking. We can best describe that change as coming from the use of mathematical models. These models are simple mathematical descriptions of physical elements. By manipulating the mathematics, rather than the physical elements themselves, one could gain a lot more insight and could design more complex systems. The older version of engineering as an artisan activity was replaced by a scientific base. The older type of thinking did not survive as a viable approach in the newer engineerings (such as electronic). If you are younger than fifty years old, you did not personally observe this revolutionary change. The *early* audio engineers were part of it.

While it is clear that there were dramatic changes in electronic engineering from the period of 1950 to 1980, the way of thinking did not change. All the talk about rapid changes in engineering during this period were changes of content. We went from vacuum tubes to transistors, to integrated circuits, to digital, to microprocessors, etc. Many engineers had trouble keeping up with these content changes because it did require the learning of a large amount of new material. Nevertheless, the process and way of thinking remained stable. One made simple models of the elements. One predicted behavior of combinations. One built proto-types in the laboratory. And one debugged and tested the result.

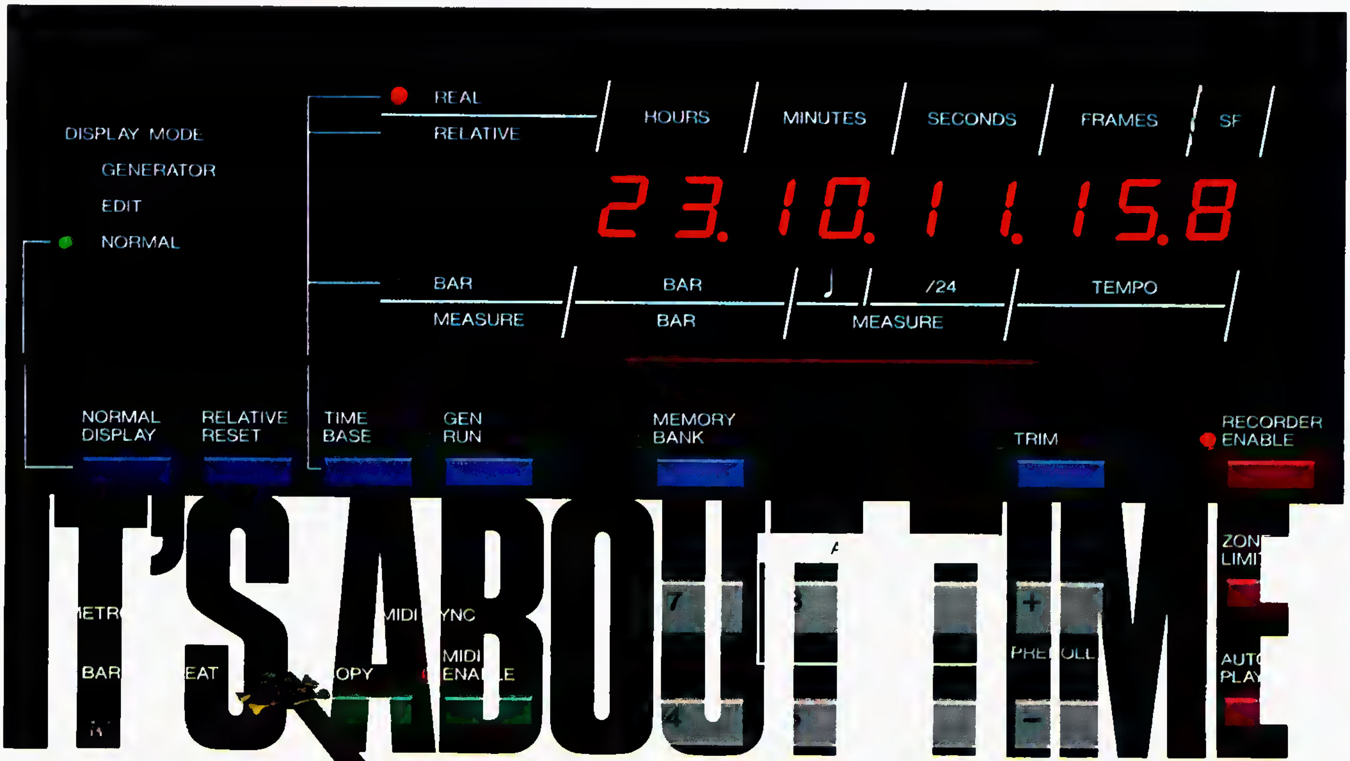
Digital audio was born in this period. Although the training and abilitiies were placing greater demands on engineers, engineering was still based on the physical properties of the elements. The ultimate harmonic distortion of an amplifier could be traced to certain characteristics of the silicon used in makng the transistors. These limitations could be overcome by clever manipulation of the design. The physicists were still the parents of the engineers.

CHANGING PARENTS

The first suggestion that the mode of engineering is about to change comes from an examination of computer programming. Computers have been the exclusive domain of electronic engineers because a knowledge of physical elements is required in order to construct such large hardware systems. Having engineers build such systems, it is obvious that the first programmers would be those engineers. Hence, programming had engineers as parents. The evolution of software, however, now shows that they were the wrong parents! When one examines the activity of writing a software program one notices that there is no requirement to understand the computer or the technology which was used to build it. The start ing point for programming is the understanding of a limited set of rules; these rules are not that different from the rules of other games such as chess.

The complexity of playing chess comes from the human ability to manipulate the set of rules; it makes no difference if the pieces are built from plastic or wood. Two computers which have the same set of rules are completely equivalent regardless of the technology used to construct the computer.

A very interesting observation can be found if one looks for the basis of defects in a computer program. All defects are human defects which come from human failures. The probability of creating a defect relates to the method by which the human programmer organizes the mental activity. With very small simple programs, many different methods of human mental processes can result in a good



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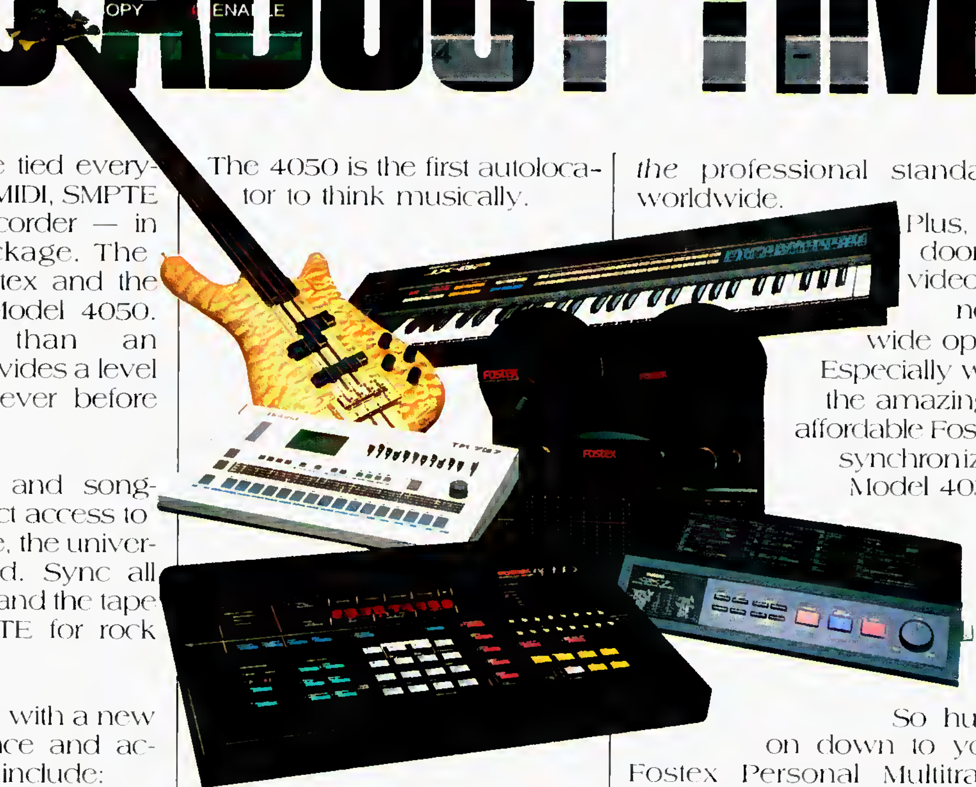

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program. When the programs become very large, the limitations of human cognition become very dramatic. How would you write a program having 30,000 lines of code? While each line of the code is easy to understand with only a high school education, the composite properties of the total program cannot be understood by any human.

I would like to illustrate this with an analogy. Suppose I give you the task of examining ten pennies and tell you to turn over those pennies which show tails. The goal is to get all pennies to show heads. It is clear that we could all do the task and we would all have a very high probability of getting that task one hundred percent correct. It would not matter how we did the task. Now I will extend the problem. Consider that you enter a room which had a floor area of thirty feet by thirty feet and that there were 10,000 pennies on the floor. How would you do the same task? The direct approach will fail. You cannot do the larger task even though it is an extension of the earlier task.

A professional programmer, recognizing the human limitations, would approach the problem in a completely different way. The first task would be to build a wooden grid above the floor to solve two problems. First, the floor area is broken up into a larger number of smaller units; secondly, one can now move about the room stepping on the wooden grid and not on the pennies themselves. This prevents one from creating an accident by hitting pennies in the process of walking. The activitiy of building the wooden grid is that of building a tool before starting the task. With this tool, one can now work on each one foot square the way one would have done with the simpler problem. The notion of building the wooden grid comes from a knowledge of human limitations not from physical properties. A robot might be able to do the task directly without such a tool.

By understanding these issues, we can see that the better choice of parents for computer programming would be cognitive psychology. That set of parents is concerned with properties of human cognition and human mental activities. Several months ago, I wrote an article in this column about my experiences with the design of custom integrated circuit using a computer system. As a hardware engineer, I had initially approached the problem in the same fashion as I had approached other hardware design activities. My parents were still the physicists. After a short time, it became clear that these were the wrong parents. The computer system had such a good model of the individual gates and the rules for combining the gates that the computer took care of all of the physical properties of the elements. It had reduced the physical world to a set of rules. This was identical to what happened with computers and software. The design function of hardware was reduced to computer programming.

If I did not make any errors, the result would be perfect. All defects were human defects. Furthermore, the debugging tools were so good that I could examine any part of the design in as much detail as I wished. An error in the design could thus come from only one of two sources: forgetting to examine a case, or examining a case and not noticing that the result was not what I had intended. In the experience, I started to follow some of the rules which computer programmers use. Never work on a module which is too large. The size is defined by my ability to hold the entire module in my head at once. Use a human readable notation for the names of signals so that the operation of the signal was clearcut and unambiguous in the name. This prevents one from having to keep the origin of the signal in one's head. Do not make the design clever. Clever designs are much harder to understand.

The activity of designing my own integrated circuit was demonstrating that I had acquired new parents. The old parents would not have helped me in the new activity. Interestingly, the content of the activity was not new. Designing digital signal processing systems was something which I had been doing for fifteen years. The difference between the old TLL logic and the new gate arrays was very small. What had changed was a dramatically new way of thinking.

AUDIO ENGINEERING

If you, as a reader, are not a design engineer, but a user of audio equipment, you may be thinking that this discussion is interesting, but not directly relevant to me. I will now show how wrong your conclusion is. We will examine the next generation of digital audio mixing consoles. In doing this, we will assume that the technology is advancing at its current speed. Signal processing devices will be reducing in price such that a console had available relatively unlimited signal processing power. It is not a very valid assumption if one reads the technical newspapers.

This means that the console could have a very large number of features. Each channel would have the ability to have compression, limiting, spectral equalization, reverberation, monitoring, pitch shifting, etc. Each of these functions in turn would have ten to twenty user adjustable perameters. Flexibilitiy and power make the result appear to be very attractive. Now comes the hooker: how can a human manipulate all of those controls? Is the task not that different from the example of 10,000 pennies? How should the state of the machine be visually and acoustically presented to the audio engineer? I believe that this is not a technical question but a human mental question. The design of the "front panel" is a non-trivial issue which does not have an engineering parentage.

Twenty years ago I visited an electronic music studio in Stockholm which was the most advanced of its day. It allowed the user to create about thirty individual instruments simultaneously. Each instrument had its own unique waveform. In order to provide the largest amount of flexibility and utility, the designers had allowed the user to set one hundred individual points of each waveform independently. The system was never used because no human could understand how to set all the controls. The fact that one could synthesize a good piano sound looked like it made the synthesizer very powerful, but no human knew what settings to enter into the controls.

In many respects, the audio engineer is better able to handle the change in the way of thinking because music has always had a degree of human artistic and non-technical basis. What will be new for the audio engineer is the visual and cognitive manipulations. A musician manipulates the acoustic result from years of practice on his instruments. Audio engineering will need similar experience.

Contrary to most of our thinking, understanding human abilities and limitations is not something that most of us have been trained to do. We think that because we can ride a bicycle we understand how to ride a bicycle. This is not true. One of the formalism which will become a new profession in the 1990s is linguistics and cognitive processes. The term "user friendly" is the beginnings of this way of thinking. Unfortunately, the properties which make something friendly or unfriendly are not yet understood.

It is now time to get to know your new parents! ■

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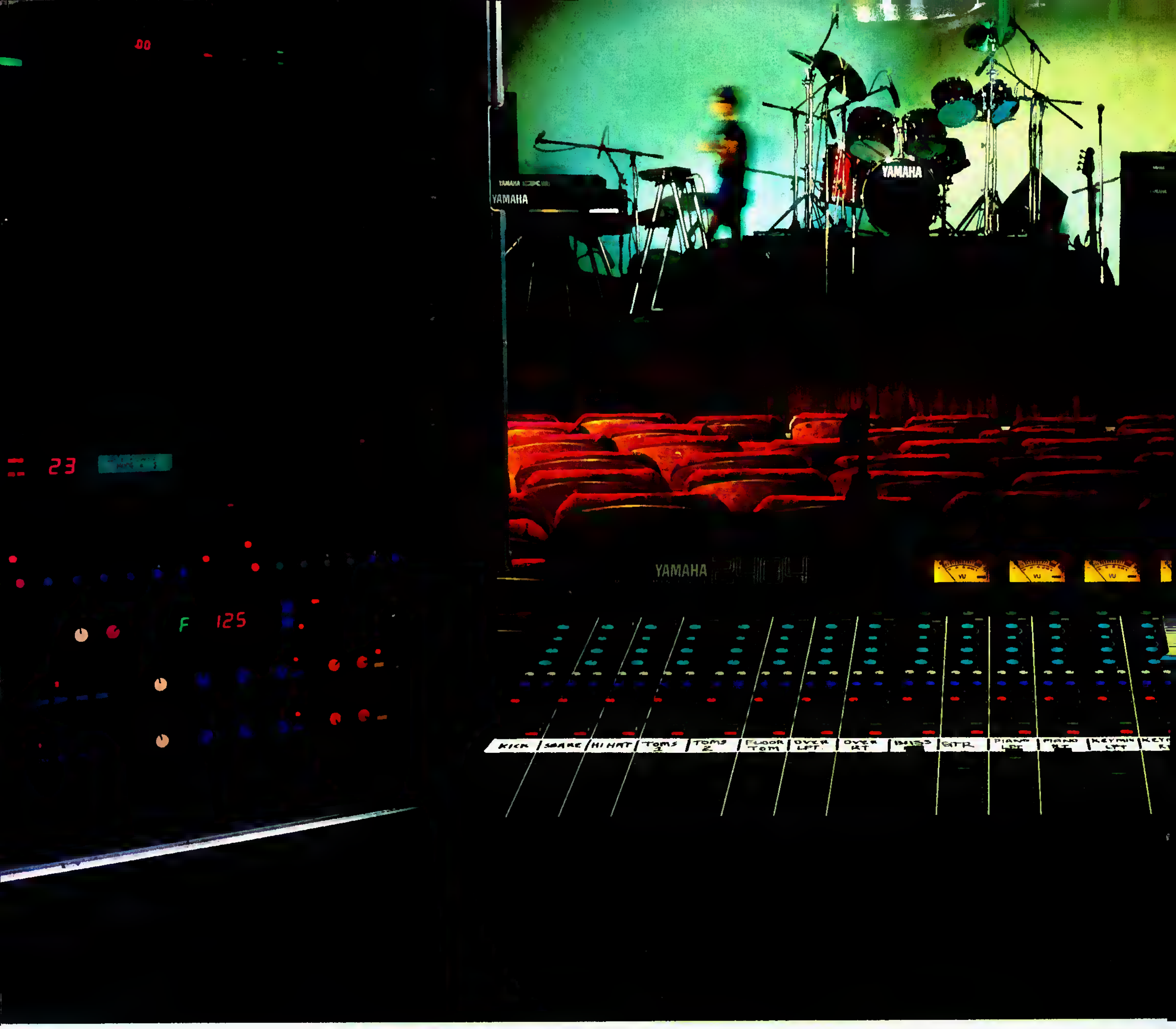

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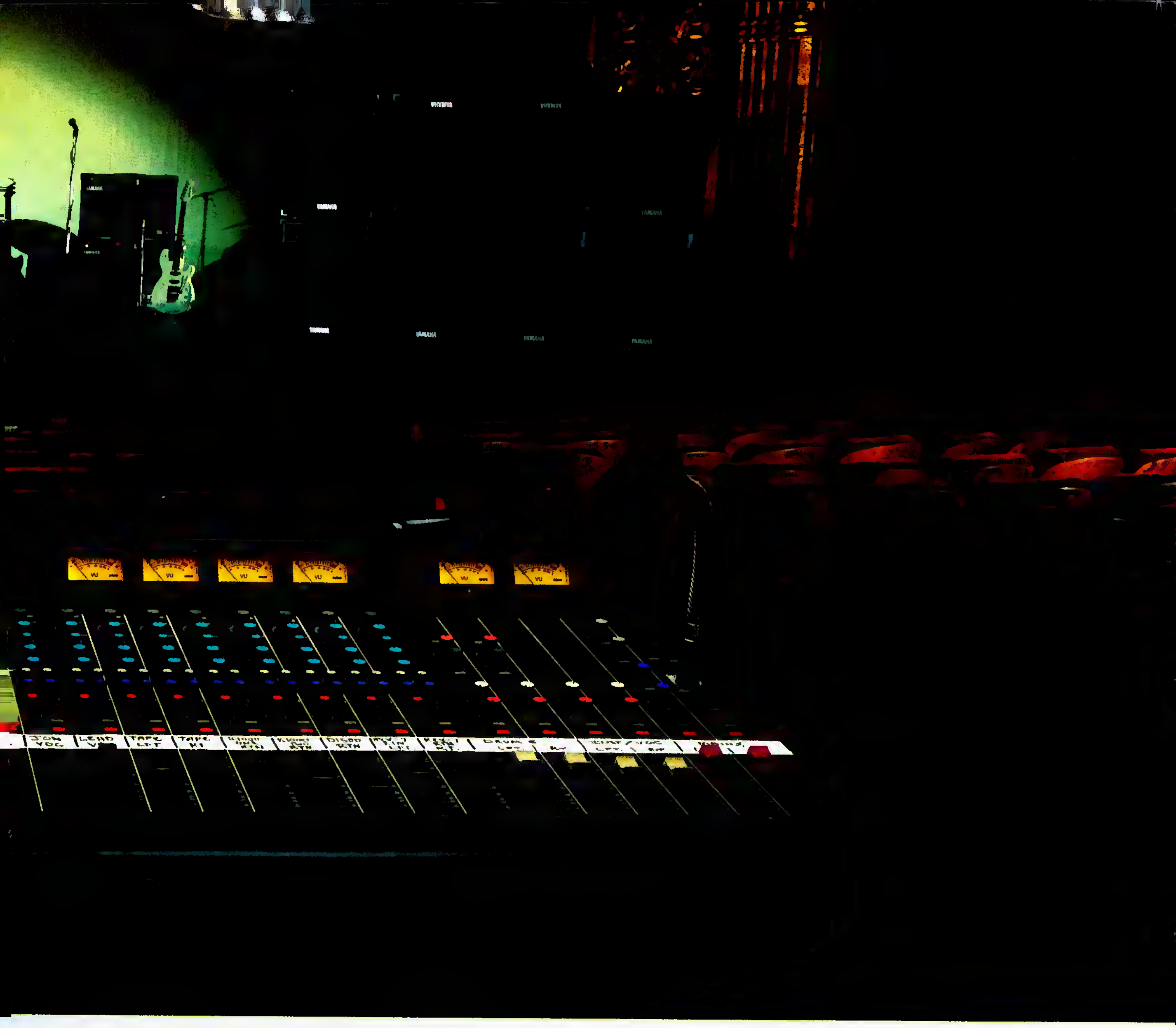

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tronically balanced with XLR-type connectors. And there are insert patch points on all input channels as well as on the groups.

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Bruce Bartlett

Recording Techniques

Common Faults In Beginners' Recordings

• Home recordings made by beginning engineers or musicians often sound dull, dead, and muddy compared to records. What causes these flaws, and how can they be corrected?

Each element in the recording chain—microphones, mic placement, mixer, effects, tape, and tape recorder—contributes to these problems. Let's focus on each element and clean it up. The end result will be better, more professional sound from your tapes.

DULL SOUND

"Dull" sounding recordings lack sparkle or sizzle. They sound like they have been played through a stereo system with the treble turned down. Cymbals sound muffled.

Dull sound is caused by high-frequency loss or roll-off. Let's examine the links of the recording chain to see how each can contribute high-frequency loss.

Microphones: If your recording microphones roll off at the high end of their frequency range, the highest overtones of the instrument being recorded may not be reproduced at their proper levels in relation to the fundamental. This high-frequency loss is perceived as a dull, muffled timbre. You need a microphone with an extended frequency response—flat up to 15 or 20 kHz—to avoid dulling the sound of cymbals, acoustic guitar, and percussion.

Such a response is usually provided only by quality condenser microphones. So if you want to record cymbals with all the crisp "ting" and sizzle of the real thing, use condenser microphones with a good high-frequency response. Check the frequency-response curve in the manufacturer's data sheet to ensure that the response extends to at least 15 kHz with little roll-off.

Note that some dynamic and ribbon mics have excellent high-frequency response. Conversely, not all condenser mics have an extended high end. Study the specs and the response curve.

With your microphones carefully chosen, you should have a bright, crisp sound right at the start of the recording chain. Elements farther down the chain, however, still can roll off the highs. Let's prevent that from happening.

Microphone Placement: Even if a microphone has an extended response, high frequencies may be lost if the mic is placed poorly in relation to the instrument. That's because high frequencies from an instrument radiate differently in different directions. We have to place the microphone in a spot where there are sufficient high frequencies.

Typically, hi-hat cymbals radiate high frequencies best from the edge of the cymbals. Placing the microphone just above the edge of the hi-hat cymbals picks up a bright sound.

Another cause of high-frequency roll-off is *microphone off-axis coloration.* While the frequency response of a mic may be flat on-axis (directly in front of the mic), it may have a rolled-off or colored response off-axis. Try to keep cymbals as on-axis to the mic as possible; aim the cymbal microphones at the cymbals. The snare-drum microphone is often used to pick up the hi-hat cymbals. Aim this mic partly toward the hi-hat to avoid dulling the hi-hat sound.

In general, the smaller the microphone diaphragm, the better the off-axis high-frequency response. Consequently, small-diameter condenser mics are the preferred choice for mic'ing cymbals.

Cassette recorder: The response of many cassette recorders is lacking at high frequencies. Many record flat only to about 12 kHz, and this response error is compounded when tracks are copied or bounced. We can't do much to recover those lost frequencies except to boost the highs during recording (say, 2 dB) or use microphones with a peak in the high-frequency response above 10 kHz.

Since bass instruments don't produce extreme high frequencies, it's wise to record them first when bouncing tracks. Multiple bounces won't degrade the high end of the bass. Record cymbals last when bouncing tracks to prevent dulling their sound.

It also helps to use Chromium tape (if the cassette deck can be biased for it), because such tape generally allows a more-extended high-frequency response than ferric tape. Sixty-minute cassette tapes are less likely to stretch and break than 90- or 120-minute tapes.

Oxide and dust shed from the tape accumulates on the tape heads, causing high-frequency loss and dropouts. Be sure to clean the heads (and the rest of the tape path) before every recording session, and let them dry for a minute before they contact the tape. Denatured alcohol and Q-tips are often used.

Demagnetize the heads with a tape-head demagnetizer after every eight hours of use. Cover the pole pieces with tape to avoid scratching the heads. Turn on the demagnetizer at least three feet away from the tape deck and approach the deck slowly. Touch the demagnetizer to a tape head and *slowly* draw it away, then repeat this procedure for each tape head and tape guide. Turn it off at least three feet from the tape deck. Keep it away from tapes!

Avoid excessive recording levels with drums and percussion, because the recorder's frequency response gradually loses high end as the recording level is increased. This is especially true of cassette recorders.

Mixer: Turn up the high-frequency equalizer (10 kHz or higher) while recording drum tracks so that the tape playback sounds as bright and crisp as the live instrument. Again, be careful not to record drums or percussion at too high a level because this will saturate the tape at high frequencies, causing distortion and loss of high end.

It's common studio practice to boost the upper midrange (say, 3 to 5 kHz) on many instruments and vocals to add punch and presence. You may want to add a few dB of boost in this area, or simply turn up the treble control on some mixers. Compare the tonal balance of your tapes (bass-midrange-treble balance) to commercial records to see if you're adding the right amount of boost.

It's better to boost high frequencies during recording than during mixdown, because boosting high's during mixdown emphasizes tape hiss. You can improve the signal-to-noise ratio of bass tracks by rolling off the bass while recording and boosting it during mixdown.

Effects: Some signal processors such as the Aphex and EXR Exciters can make recordings *sound* brighter without affecting the high-frequency response. Exciters work by high-pass filtering the input signal, distorting it with even-order harmonic distortion and phase shift, and blending the filtered, distorted signal with the main signal. The processed signal is mixed in about 20 dB below the main signal. If your tapes sound dull, try adding some exciter effects to restore the lost sparkle.

DEAD SOUND

"Dead" sounding recordings lack spaciousness or "air"—as if they were recorded in a linen closet. The problem is a lack of *reverberation*. This is a pattern of sound reflections (echoes) off the walls, ceiling and floor that becomes more dense with time and gradually decays. Reverberation is the sound you hear just after you shout in an empty gymnasium or cathedral. Reverb is a cue to the ears that you're listening in a large room; hence, reverb adds a sense of space.

To add reverberation to your recordings, record the instrument or vocal in a bathroom from at least one foot away. Or use an artificial reverb unit: a device using a spring, steel plate, or digital delay to simulate room reverberation. Many guitar amps have built-in reverb.

Not all instruments require reverb: kick drum and bass are usually recorded without reverberation to keep their sounds "tight" or well-damped.

Mixing down in stereo adds spaciousness. In the July '85 issue of *MR&M* an article by Bob Buontempo explained how to create stereo mixdowns with 4-track recorders.

MUDDY SOUND: EXCESSIVE BASS

"Muddy" or "boomy" sound is too bassy. Excessive bass on vocal tracks usually is caused by mic'ing too close with cardioid microphones. Most of these mics boost the bass when used up close—a phenomenon called "proximity effect."

You've seen vocalists on TV singing with their lips touching the microphone grille screen. Many beginning recordists think that's the way to record vocalists. It's not. Keep the vocal microphone at least eight inches away from the singer to avoid bass buildup. Or turn down the bass on your mixer until the sound is natural.

Incidentally, it's a good idea to place the microphone above the vocalist's mouth and use a foam pop filter to prevent explosive breath noises ("pops").

An alternative is to record with an *omnidirectional* microphone because it does not exhibit proximity effect. With omnis, you can mic as close as you want, yet maintain a natural tone quality without boomy bass. Some cardioid microphones work on a "variable-D" or "multiple-D" principle which reduces proximity effect.

Another cause of bass buildup is a subtly boosted low-frequency response in the cassette recorder. If your tape playback sounds bassier than the signal going into the recorder, the recorder is emphasizing the low frequencies. This frequency-response error is doubled every time you bounce or copy tracks. If this is occurring, roll off the bass slightly on all tracks during mixdown.

MUDDY SOUND:EXCESSIVE LEAKAGE

"Muddy" also means "unclear." Lack of clarity might be caused by *leakage* between microphones—pickup of off-mic sound. Leakage occurs when, for example, the drums are picked up by the acoustic-guitar microphone. The sound from the drums "leaks" into the mic meant for the guitar. When you monitor the drum mics alone, the drums should sound tight and clean. But they may become muddy or distant-sounding when you bring up the guitar microphone (which picks up the drums at a distance).

There are at least three ways to reduce leakage: (1) mic close with cardioid microphones, (2) overdub quiet instruments and vocals, and (3) record bass and keyboards direct (with a direct box).

Mic'ing close reduces leakage as follows: As you place a microphone closer to an instrument, that instrument is picked up more loudly. But the level of background noise (leakage) stays constant with mic'ing distance. Consequently, close mic'ing lets you record a high ratio of wanted-to-unwanted sound.

Overdubbing eliminates leakage because only one instrument or group of instruments is recorded at a time during an overdub. Professional studios overdub vocals as a matter of course.

Direct boxes can be used on electric instruments in place of microphones to eliminate leakage. With a direct box, the signal is picked up directly off the instrument or its amplifier. You eliminate the microphone and its leakage pickup.

Muddy sound also is caused by excessive reverb; use it sparingly.

SUMMARY

We've suggested a number of ways to improve the sound of recordings:

- On cymbals and hi-hats, use small-diameter condenser microphones with an extended high-frequency response.
- Aim the cymbal microphones toward the cymbals, near their edges.
- To compensate for recorder high-frequency loss, use bright-sounding microphones (with a high-frequency rise) or boost the highs on your mixer during recording.
- Clean and demagnetize the tape heads, etc., before every session.
- Use high-quality tape (Chromium if possible).
- Avoid excessive recording levels, especially with percussive sounds.
- Boost the upper midrange slightly during recording to add punch and presence.
- Use exciters.
- Add reverb by mic'ing at a distance in a hard-walled room, or by using an artificial reverberation device. Apply reverb sparingly. Don't add reverb to kick drum or bass. Mix down in stereo.
- To prevent excessive bass: Mic vocals at least eight inches away (during overdubs), or roll off the bass on your mixer for close-mic'ed vocals, or use omnidirectional microphones.
- If the recorder boosts the low frequencies, roll off the excess bass during mixdown.
- To reduce leakage, mic instruments closely. Roll off the excess bass if cardioid microphones are used.
- Use direct boxes.
- Overdub quiet instruments and vocals.

If beginners would try all these suggestions, their tapes should be markedly improved—brighter, clearer, and more spacious. ■

2 to 8 trk

Stephen Cullo

Composing The Action

Let's take a look at a small studio in New York which is primarily used for recording theme music for sports.

MY SPECIALTY has been composing and recording the theme music of sports programs for the past three years. The music has been broadcasted on shows produced by Main Events of Totowa, New Jersey, (including the Pryor/Arguello fight in Las Vegas), Don King's boxing shows, and Katz Sports College Football. I'm currently working on Sporty Bear Productions' "Met Golf 86" show, a monthly golf magazine series. One thing that surprises many people is that all of my recordings are done using a Portastudio. While the four-track cassette is one of the smallest recording formats available, it continues to be widely used for demos and now, more often for a finished product.

The first thing I purchased was the Portastudio because it was affordable and had already proved that it would be around for a while. I was using a Prophet 600 and Oberheim DX drum machine along with my guitars and basses. At that time I would borrow a spring reverb for mixing to my ReVox A77 two-track. The first major change in my sound came when I added the Yamaha DX-7. Although electric bass is still my main performing instrument, the DX-7 allowed me to explore composing using the keyboard, and sparked my interest in synthesis in general. Now, I like to compose and play all the parts so the DX-7's variety of sounds is a big help. Another thing I like about the DX series (besides the fact that most people know and like the sounds) is that it has an extensive user network of patches and software. One way I've found Yamaha to be very helpful is in having a toll free number to call and talk to technicians if necessary. I also added a DX100 which is a scaled down version of the DX-7 that still provides all of its characteristic digital sounds.

All of my analog sounds come from an Octave Plateau Voyetra Eight. It's one of the most sophisticated synthesizers made and has a variety of sounds that I've never heard on any other synth. One really nice feature is that it gets periodic software updates free from the factory. They're constantly adding more functions to it and making it usable with more types of synthesizers and computers.

For percussion, the Yamaha RX11 has good basic acoustic drum sounds and, at the time of purchase, had more MIDI functions than most drum machines. I have since added the Roland TR 505 because of its excellent percussion and great tom sounds. To round out the percussion section is the Roland Octapad.

Stephen Cullo has been composing and recording theme music for sporting events for the last three years.

All photos by Michael Marzelli.

Equipment List

Tascam 244 Portastudio

Revox A77 1/4-track

Yamaha C200 cassette

Korg SQD-1 Sequencer

Yamaha Rev-7 Reverb

Yamaha D1500 Delay

Roland CE-300 Chorus

Boss BX-800 Mixer

Ibanez DM 1000 Delay

Yamaha NS-10M Monitors

Scholz Rockman

Shure mics

Various pedals

Ibanez UE405 Effects Rack

OEI Voyetra Eight

Yamaha DX-7

Yamaha DX100

Roland TR505 Drum Machine

Yamaha RX11 Drum Machine

Fender Jazz Bass w/EMG's

Steinberger XL2 Bass

Yamaha 5 String Bass

Gibson Les Paul

Ibanez RS 440

Roland Octapads

IBM XT Personal Computer

OEI Sequencer Plus Software

To bring things up to date, I believe the main factors that account for great recordings, in any format but especially smaller ones, are MIDI, the use of computers for sequencing, and more affordable professional digital processing. I started using a computer for sequencing as soon as Octave Plateau's Sequencer Plus software package came out. This program for the IBM PC has powerful editing capabilities: punch in/punch out, adding moving or deleting any note or note value on any track with complete dynamic control. I don't find it necessary to use the quantizing that much. I prefer to record a straight performance and be able to correct minor errors or add complex parts to give a piece a natural feel. Working this way can save hours of actual performance time. Another basic use of the sequencer (both the IBM PC and Korg SQD-1) is to devote a track to patch changes alone. This way one synth can be used for several different sounds throughout a piece, better utilizing track space in the four-track format. This is especially useful when I'm going right to two-track and using the Portastudio as a mixer only.

By using serveral patches on each synth I can have more parts per channel. I can also have the computer change the programs on the Yamaha Rev-7 digital reverb and D1500 digital delay to correspond with the patch changes. If I do go to tape on the Portastudio, I can use up to eight inputs on the Boss mixer and go stereo to tracks one and two. This leaves tracks three and four open for solos and/or vocals. That's ten tracks (eight pre-mixed) without any bouncing! Also, using the tape access loops on the Portastudio allows you to use four different effects in addition to the Rev-7 which I use as the main stereo reverb. These are some of the simpler setups, so you can imagine what's possible if you want to get elaborate. I also greatly improved the sound of my mixes by getting a pair of Yamaha NS-10M monitors.

The total cost of my studio up until now has been roughly $20,000. I buy the majority of my equipment from Manny's on 48th Street in New York. Besides having everything, they're especially helpful to me. Rick Stevenson has been particularly helpful in assisting me with my choice of equipment, programming, and keeping me informed about new toys to buy.

Since I like to play most of the instruments myself, I receive a great deal of assistance in computing and engineering from Andrea Bella. I also use my brother, Chris Cullo, for drum programming and percussion parts.

What I'd like to do in the future is continue with the sports pieces and do more jingle and soundtrack work (both film and TV). I really like the music on TV's *Simon & Simon* series and recent Honda commercials. It definitely shows how creative you can be. This fall I'm moving my studio to a loft in Chelsea. Part of my expansion will include some type of SMPTE synchronization for video/film. Other improvements I'd like to make would include an eight-track tape deck with an Allen & Heath CMC mixing board and some type of sampling module.

I basically enjoy writing and recording music on any level so it's fun even with a Portastudio. However, I do look forward to doing everything on a grander scale in the future. ■

Mark E. Battersby

On Taxes

Basic Bookkeeping And Records

• When all is said and done, there are really only two reasons for a home recording studio owner to keep records of its operation. They are required by law and they are extremely useful in managing the studio as a business. Of course, there is also such a thing as too many records or too much bookkeeping, so how is the average home recording studio owner to know what happy medium to aim for?

A good way to determine what type of records are required for a particular studio operation is to take a closer look at what records are utilized by others with similar operations—as well as considering what records and information are actually needed to run the recording operation for maximum profit.

Obviously, records must be kept to help determine the tax liablility of the studio or recording business. Regardless of the specific type of bookkeeping system employed, our tax laws require that the records must be permanent, accurate, and complete. Plus, those records must clearly establish income, deductions, credits, and employee information, as well as anything else specified by federal, state, or local regulations. Remember, however, that the law does not require any particular kind of records, only that they be complete and separate for each business.

At the outset, the type and arrangement of books and records suitable to a particular studio operation should be established, keeping in mind the various taxes for which the business is liable and the times at which they are due. If this is not an area in which the studio owner feels competent, outside professional help is called for.

Setting up a system for good record-keeping need be done only once; doing it efficiently makes things much easier later on. Thus, there is no real need for professional accounting help beyond the setting up of books. If loans or local operating rules require financial statements or regular audits, accounting assistance may be necessary on a regular basis, but never purchase more outside professional aid than is really required—or than will be used.

Double-entry bookkeeping is usually the preferred method for keeping business records, making use of journals and ledgers. Transactions are normally entered first in a journal, and then monthly totals of the transactions are transferred or posted to the appropriate ledger accounts. Ordinarily, the ledger accounts include five categories: 1) income; 2) expenses; 3)assets; 4)liabilities; and 5) net worth. Income and expense accounts are totalled or closed each year. Asset, liability, and net worth accounts are maintained on a permanent and continuing basis.

Many smaller studios discovered long ago that the two required steps in double-entry bookkeeping are both time consuming and complicated. One result has been an increase in simplified bookkeeping systems and the so-called "one write" checkbook systems that permit a business to write a check while at the same time posting the amount of that check to the proper expense account.

Single-entry bookkeeping, while obviously not as complete as the double-entry method, may be used effectively—and legally—in the small recording operation, especially during the early years. The single-entry systems can be relatively simple, regardless of its form, recording the flow of income and expenses through a daily summary of cash receipts, a monthly summary of receipts and a monthly disbursements journal (such as a check book). This system is usually entirely adequate for the tax purposes of a small recording business.

It goes without saying that all receipts from the recording operation should be deposited in a separate bank account with, perhaps, a petty cash fund established for small everyday cash expenses. Naturally, all business expenses paid in cash should be supported by documents or receipts that clearly show that the expenses were incurred for business purposes.

Similarly, all disbursements should be made by check if at all possible so that business expenses can be well documented. If a cash payment is necessary, a receipt for the payment, or at least an explanation of it, should be included in the business records.

Cancelled checks, paid bills, duplicate deposit slips, and all other documents that substantiate the entries made in the business records should be filed in an orderly manner and stored in a safe place. In fact, accounts should be classified in groups relating to income, expenses, assets, liabilities, and net worth. Furthermore, asset accounts should be classified as current or fixed and recorded in detail the date of the acquisition, cost, depreciation and any other items affecting the account.

Obviously, every studio owner should carefully preserve all underlying business papers. For instance, all purchase invoices, receiving reports, copies of the sales slips, invoices, or statements sent to customers, all cancelled checks, all receipts for cash paid out, etc., must be meticulously retained. They are not only essential to maintaining good records, but may also be important if legal or tax questions ever arise involving one of these items.

Payroll records present another set of problems. An employer—regardless of the number of employees—is required to maintain all records pertaining to payroll taxes (income tax withholding, Social Security and federal unemployment tax) for at least four years after the tax becomes due or is paid, whichever is later.

The payroll for a small home or small recording studio is usually a relatively simple task if the owner uses a good pegboard or "one-write" system. Any office supply store provides samples that are available.

Overall, there are over twenty different types of employment records that must be maintained just to satisfy federal recordkeeping requirements. For example, the records that relate only to income tax withholding would include:

- The name, address, and Social Security numbers of each employee;
- The amount and date of each payment of compensation;
- Amount of wages subject to withholding in each payment;
- The amount of withholding tax collected from each payment;
- Any reason that the taxable amount is less than total payments;
- Any statements relating to employees' non-resident status;
- The market value and date of non-cash compensation;
- All pertinent information about payments made under sick-pay plans;
- The employees' withholding exemption certificates;
- Agreements regarding the voluntary withholding of extra cash;
- Dates and payments to employees for non-business services;
- Statements of tips received by employees; and
- Requests for different computation of withholding taxes.

Records pertaining to Social Security (FICA) taxes always include:

- The amount of each payment subject to FICA tax;
- The amount and date of FICA tax collected from each payment; and
- An explanation of the difference, if any.

Records relating to unemployment taxes (both state and federal) are:

- The total amount of salaries and wages paid during the calendar year;
- Amount subject to unemployment tax;
- Amount of contributions paid into the state unemployment fund; and
- Any other information requested on the unemployment tax return.

Obviously, there are other records that a studio owner should maintain—and retain. For instance, records supporting the items on federal tax returns should be kept until the statute of limitations (seven years, although audits are rarely conducted after three years) expire.

Records relating to depreciable property should be retained for as long as they are useful in determining the cost basis of the original or replacement property. Copies of the federal income tax return should be kept forever according to some experts. In fact, perhaps those tax returns should really be kept forever—they may be extremely helpful to the executor of the estate of a deceased business person.

Finally, the accumulation—and utilization—of these books and records would be greatly impaired if they were lost, stolen, or damaged as a result of storing them on the business premises. Protecting the bookkeeping records used daily on the recording studio's premises from theft, fire, or other damage insures their safety. However, complete removal for safekeeping at a different location of all records not needed in the daily operation of the business makes even more sense and may help the business to be rebuilt if it ever suffers severe damage or destruction. ■

Rita Wolcott

The Professional Audio Store

In our March/April issue *db* featured an article on buying professional audio equipment through direct mail services. As was our original intention, this month we are featuring an examination of the *most* popular method of buying professional audio equipment—the pro audio retail store. *db* spoke with Martin Music Audio/Video Corp., Sam Ash Stores, and Sound Genesis, three major professional audio stores in New York and San Francisco, to get an idea of how retail audio dealers view the ever changing market of audio equipment.

A LITTLE BACKGROUND

New York's Martin Audio Video Corp.'s new operating division, Martin Music Technologies, is a full service sales organization for high end analog and digital synthesizer equipment, sampling keyboard systems, and drum machines as well as personal computer software for musical applications. They also offer a full stock of professional audio products.

Sam Ash Stores have six branches in or around New York City with the Manhattan branch divided into two separate buildings. One is devoted to electronics and the other is devoted to electric guitars, amplifiers, and drums.

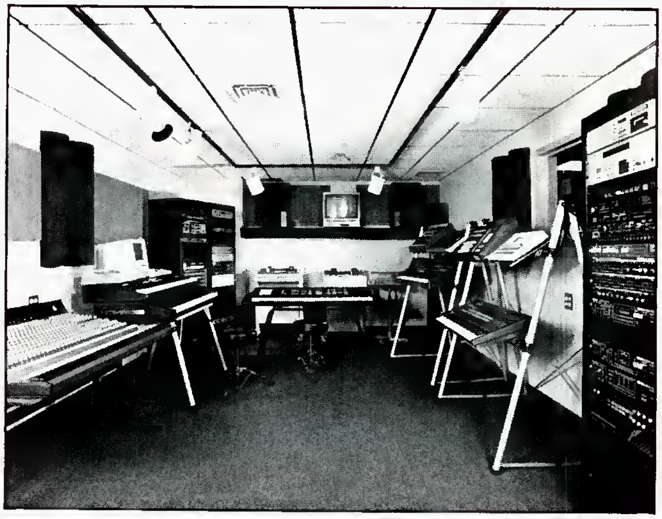
San Francisco's Sound Genesis takes a different approach and primarily services audio for video customers. They have been in business since 1968 and handle over one hundred lines of products with primary lines being Otari tape recorders, and Sound Workshop, Harris, Trident, Soundcraft, and Auditronix mixing consoles. Sound Genesis has been actively selling equipment since the mid-seventies, and was a design and service firm previous to that.

WHO'S INVOLVED?

Who is the principal customer of the professional audio retailer? Most basically serve the same type of customer with a few variations. For example, Courtney Spencer, vice president and general manager of Martin Audio/Video Corp. says, "Principally, the self-contained recording operation, the producer/composer type who is increasingly doing more creating and producing in his own studio using keyboards, drum machines, and in many cases, fairly sophisticated recording equipment. Also, traditional recording studios, many of whom are getting involved with keyboards as a major part of their activities."

Sound Genesis takes a slightly different angle. David Angress, vice president of Sound Genesis, explains, "Initially, we started out selling to the music recording studios, but as the marketplace has evolved we have been doing more and more broadcast work for both radio and television. We work with corporate recording studios which are primarily video oriented. In effect, my main consumers are recording studios, broadcast facilities, and corporate recording studios. Basically, we define anybody who uses audio equipment as a tool of their trade as a customer. The difference between that and the typical consumer is that if something breaks in the consumer's home it's an inconvenience; if something breaks for one of our customers it's costing them real money...so we're very sensitive to that."

Paul Ash of Sam Ash Stores reports, "We have a lot of facets of the market covered. Originally, we sold to musicians, but many of those musicians grew up and became professional audio engineers, producers, and people who need expensive professional gear, so we graduated, too. We sell to musicians, home hobbyists, and studios."



Facilities at Martin Music Technologies.

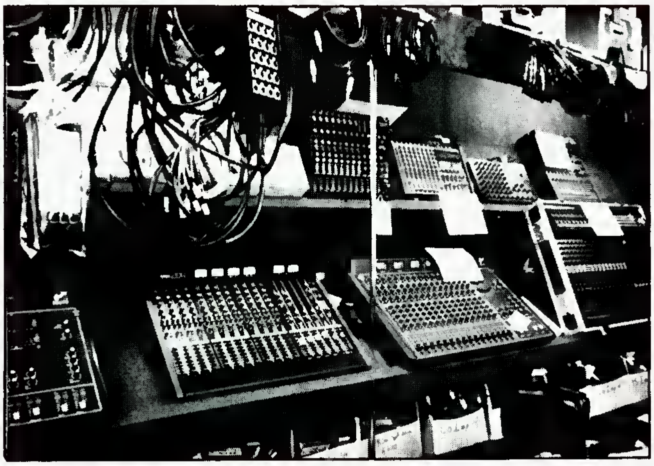
There seems to be a fine line between the professional and the hobbyist in the minds of professional audio dealers, and, at times that line has become hazy due to a surge in smaller home recording studios. David Angress offers, "It depends on how you define the smaller recording studio. In the mid-seventies, people started putting studios in their homes for their own creative use. That market was called the semi-pro market. I've never really liked that phrase, but we use it because everybody seems to know what it means. I don't know...you're either a professional or an amateur to me. I've always had a hard time with semi-pro."

Paul Ash reports another customer, the home hobbyist, as someone who does not intend to take his work into a larger studio. "There's a whole group of adults who are hobbyists. They're not really professionals and their own playing has nowhere to go. When they reach a certain age and they're not in a band and they're not professionals, years ago they would pack the instrument away in a closet and that would be the end of it. But now if you have home recording equipment like even a small multi-track, you can put one track down...you buy a little Casio keyboard, you buy a drum machine, and you can do a multi-track recording."

THE NEED

What's behind this home recording studio boom? Obviously, in the last five years there has been significant market growth resulting from a growing interest in the smaller recording studio. The reason, of course, lies within the development of synthesizers and MIDI, both of which will definitely continue to play an increasingly important role in the future. Consequently, as equipment has become better and better and its users have become more technically oriented, there have been more finished products actually being done in the home or smaller recording studio.

Paul Ash agrees, "The best way to go into an expensive studio is to be prepared at home. You *can* do a professional recording job like Bruce Springsteen did on *Nebraska* on home equipment. More and more people are realizing that." And that's not to say that work being done in a home studio is work considered less professional in the long run. Paul Ash continues, "At this point there's a lot being done in individually run studios. Although they're certainly smaller than some of the big studios, they aren't necessarily less competent in terms of the technical level of equipment."



Facilities at Sam Ash Stores.

Audio for video has also emerged as a very strong market in the last few years, and with it pro audio dealers have been busy taking television stations into the world of stereo by transmitting the signal into stereo. David Angress adds, "It also means replacing audio equipment that was very good when it was made, but just doesn't have the specifications required to deliver high quality sound for the video market."

MIDI *has* brought us far in a relatively short period of time, yet it doesn't seem to be a completely overwhelming force—take a look at live recording for example. The commercial music business in particular is still very much involved with live bands, real drummers, and so forth. The same holds true for album work and film scoring. A ready conclusion may be that it's a sharing of the spotlight as opposed to the grabbing of it. Courtney Spencer says, "MIDI-type studios certainly will become more important as the technology becomes more developed, but multi-track recorders will be with us for a long time—analog or digital or disc based or whatever the technology develops into."

HANDLING THE GROWTH

Because MIDI, sophisticated synthesizers, and digital sampling devices have come so far and are in such demand, national retail stores have had to scramble to keep up. How are they handling it? New divisions and expansions like Martin Music Technologies have been one answer.

Courtney Spencer offers *his* reasons for the new division. "We felt that the specifications in terms of putting systems together were such that we might be able to offer something unique by proceeding customers through the various steps in order to create a showroom store kind of environment. This is the type of place where people with smaller recording products like 1/2-inch 8-track compatible mixers and signal processor products would have a more familiar and comfortable environment than the traditional Martin Audio sales office situation which is what we've worked with for many years. We really tried to kill a few birds with one stone, like covering the MIDI technology area while creating a show place for some of the more affordable recording equipment."

Sam Ash was traditionally a musician-oriented store, but in the last few years, it too, has grown with the times. Paul Ash comments, "I understand that most of the acoustic instrument companies aren't doing too well, yet we're having the best year in our history. We're up about thirty percent over last year and it's because we're into the technical end—keyboards, MIDI equipment, recording equipment, and professional sound equipment. In the last five years we've gone more into the high end technical stuff so we've been reaching up and now we've got 24-track equipment. A lot of the more expensive sophisticated equipment is much cheaper than it used to be. Now you can get great synthesizers for $1,000 that would have cost $4,000 a few years ago."

CHANGING PATTERNS

It certainly looks as though the self-contained studio has become more and more important. The large central studio, as far as the total slice of the pie, has been the commercial studio for hire. It has remained a constant factor in the business, but a lot of growth has developed out of the privately owned facilities. Keyboard products, MIDI products, drum machines, and so forth have become increasingly important over the last two or three years. Signal processing has also become very hot and digital is really starting to take hold, particularly with the more affluent producers and studios.

And because video is very much a large part of today's market, it too, has captured the attention of audio experts. More and more of the equipment being sold is synchronized to video. Consequently, the requirements for very stable transports that can be easily controlled by a time code based synchronizer have become important. David Angress states, "The synchronizers themselves have not only been getting better, but have been getting easier for an engineer to operate and are significantly more reliable. In the mid-seventies all of our customers started jumping from 4- to 8-track. Essentially, everyone now wants a professional 24-track or the 1-inch 16-track, which is also extremely strong. Otari has a series called the MX-70 that is doing very well. A lot of those machines are finding their way into keyboard oriented studios."

WHO KNOWS WHAT?

With a rapid growth in technology often comes a confused layman who must learn how to properly operate highly sophisticated equipment. However, today it seems that most users have not only caught on, but have practically become experts in the field. Courtney Spencer has several firm ideas on the subject. "I think that a lot of the composer/musician/producer types are bright people and are very into technology and they've gotten themselves to be knowledgeable quickly because they have become quite sophisticated along with the whole realm of technology."

Paul Ash adds, "When synthesizers first came out we were the fist store in New York to sell them. We held classes on what a synthesizer is and what synthesis is, and we had classes for school teachers who went out and taught it and, of course, for musicians. Now that's unheard of! People come in and kids seem to be *born* knowing about synthesizers. Yet, I feel there is always a need for more and more information and that's why I found that the weekly seminars I hold are going so well. We talk about reinforcement, computer software, multi-track programming of the DX-7—things like that."

SALES BACKGROUND

What else do the major retail dealers do to keep abreast of the latest advances in technology? Well, that is often very much up to the sales force—or as dealers generally call them—the *service* staff. Today the pro audio salesperson is expected to be typical of the users, with significant amounts of studio and/or broadcast experience. In essence, they need to know as much as, if not more than, the customer. The typical sales person in today's retail store is a musician or an engineer, not someone fresh from business school. Paul Ash points out, "There are some musicians who are more interested in the music and then there are others who are more interested in the equipment. They call themselves equipment junkies and that's what we need—people who are more concerned with the equipment and know how to talk about it and use it. They don't have to be slick salesmen; they have to know and love the equipment."

David Angress agrees, "Actually, hiring is one of the most difficult situations here because we need people who not only have a technical background, but also have an applications background as well so they understand how the client is going to be using the equipment."

WHEN SOMETHING GOES WRONG

Most audio dealers handle repairs either by servicing the equipment in their own repair centers or shipping them out to manufacturers or manufacturer authorized repair centers for the customer. Courtney Spencer echos the majority when he says, "We have highly trained technicians, and in terms of our traditional markets, service has been our greatest strength. It's the thing into which we've put the most amount of attention."

In effect, service *is* still one of the major reasons that professional audio stores are still by far *the* most utilized method of purchasing equipment. For example, Douglas Wood, professional audio/video sales at Martin says, "Look at computer software. There are dozens of products designed to run a number of different computers to do various things involving MIDI. It is very hard for the customer to get a sense of whether this product is right for him either from an ad or even from just reading a review. It's easier to come in and get hands-on experience."

SUPPLY AND DEMAND

Another selling point for retail stores is that problem of ordering equipment—it isn't the consumer's. In general, about seventy-five percent of products ordered are reportedly available. Twenty-five percent are a bit more difficult to obtain. This may be due to strong demand or simply because the product may be new. David Angress uncovers an interesting aspect of buying patterns. "A lot of equipment that is imported, particularly from England and Japan, have had significant price increases in the last months with the function of the dollar going down relative to other currencies, and so whenever you have impending price increases and people realize they're coming and they were about to make a purchase anyway they just advance it a little bit. Business has been great for the last couple of months, but that puts a lot of strain on the manufacturer's ability to keep inventory in stock so popular items that come from Japan and England seem to be a little harder to get."

Sound Genesis.

TRAVELING SHOW

Paul Ash has conceptualized a solution to continuing consumer education. He suggests a traveling consumer show as a way for manufacturers to get their information out to the public. He says, "There's no reason why we can't have a good consumer show that travels. It would open up every market. When people see what's available, it whets their appetites. They may not buy on the spot, but every dealer will do better if the public knows what's out there."

CONCLUSION

It appears that the increasingly important and rapid devolopment of synthesis and MIDI has had a strong hand in changing the face of the recording industry in the last several years. With that change has come a widespread need for professional audio dealers to grow along with the small recording studio market. This has been accomplished by expanding their lines, adding new divisions, and updating their sales and service departments. It all points to big business for dealers and easier accessibility to the consumer, whether it be home hobbyist, smaller recording studio, or the large recording studio operator. ■

Ad Ventures

Brian Battles

It's Not All Suntan Oil

• Summer weather really makes it tough to buckle down and get any work accomplished properly, such as twisting knobs or even cutting the grass, because deep down inside your soul is screaming, "Party! Beer! Nude beaches! Loud music! Swimming pools! Nude beaches! Wine coolers! Softball! T-shirts! This year let's really go find a nude beach! Barbecued ribs and chicken! Frisbees! Why aren't there any damn nude beaches near here?!"

It takes a lot of stern discipline and honest commitment to concentrate on making a living, no matter what your occupation, but especially if you just happen to run a recording studio. That's because most of your clients are going to be drinking beer, eating barbecued chicken wings, and playing softball on their own private nude beach by the same time tomorrow afternoon, while you'll be up to your navel in a lap full of chapped strands of magnetic tape, breathing the stench emmitted by soundproof carpeted walls that retain every molecule of tobacco, sweat, and pizza fumes that have been trapped in their fibers since your first session.

Success, however, comes to those who toil and struggle while others enjoy their revels (recall Aesop's ant and grasshopper). So who's going to have the last laugh? I'll tell you who: the people who are partying now while their stomachs are flat and their hair is dark. Diligent, conscientious folks like you and me may someday be able to own the Porsches and underground pools and fancy foreign appetizers and a private jet to take us to Rio where virtually every beach is a nude beach. Of course, by then we'll be seventy-eight years old, incontinent, toothless, and covered with liver spots. (You wait—it'll happen to you. Yes, you, Spa-breath! Turning into a walking medical encyclopedia is the insidious and inherent fate of our "hoping to increase their lifespan" generations). Fortunately, there will always be a supply of healthy, nubile, attractive young members of the opposite sex who will hang around with you anyway if you have all of the things rich people seem to have.

If I sound a trifle cranky here, it's only because I wrote this on a beautiful, warm but not too muggy summer evening in my New England home, sitting in the upstairs den with a sweating glass of Kool-Aid inches from my keyboard. This was in sharp contrast to my unfulfilled fantasy of lying on a soft blanket on a nude beach with a beer in one hand and suntan oil in the other. (*Note To The Editors: Yes, poor Battles finally bought a word processor. Next, he is going to scrape up his pennies to purchase a modem so he won't have to buy copy paper.*) No more evading the issue; I suppose I really should earn my keep by jotting down something about recording in here, so let's take a deep breath and plunge in.

As I mentioned last time (and I know you memorized the entire column), you can make an awful lot of extra money by using your recording facility to produce radio commercials. In fact, many studios do all or most of their work putting together broadcast jingles. Obviously there's money to be made. Add to that the fact that the market is still very much wide open, and you're talking serious dollar potential. I'm honestly wondering when some entrepreneur is going to open a chain of homogenous commercial factories with studios in all major cities, marketed directly to small and medium sized businesses (McJingle's?). There already are companies that cut "generic" pieces and will overdub a client's name in the appropriate spot for a nominal fee. I've even heard this done in overlapping radio markets where alert listeners must wonder about the "coincidence" that two different car dealers have ads that sound so similar!

Summer is traditionally a slow season for many types or retail businesses, but don't think that means you can't get anything accomplished until the back-to-school and Labor Day sales kick off the big autumn/holiday campaigns. Now is the time to start building a demo tape, design stationery, scan the yellow pages, maybe assemble a neat little sales folder (containing your brochure, list of services, base prices, business card, and so forth), and do your all-important market research.

This last activity may sound dull, and perhaps conjure up images of poring over dusty Thomas' volumes in the local library, but that's not necessary yet. At this stage your market research is as simple as lounging in the sun with a portable AM/FM radio and a notepad. You simply jot down names of local sponsors on stations in your nearby cities and make notes of what kind of radio commercials they are using. Are the same voices on all the ads on one station? That means the sponsors allow the station to produce the spots, drawing from a limited pool of voices, namely the disc jockeys. Mark down these potential clients. Are the musical backgrounds bland instrumentals, or even familar recordings? If an ad is read over Herb Alperts "Rise" or Ronnie Laws "Always There," chances are you may have a customer.

As I have said before, most local radio advertisers don't know what they are really doing with the time they buy, so they trust the "experts" at the radio stations to put the commercials together for them. Even if they occasionally wonder how those customized musical extravaganzas are made, they probably imagine that it costs more money than they'll ever see. Your job is to educate and communicate. You can do it personally, you can do it well, and you can do it inexpensively. You train yourself to convince the sponsor that not only is your service affordable and professional, it is vital. As long as they are laying out the money for the time, why not pack the most punch into that sixty second rental of ether?

In order to have your presentation seriously considered by a client, you must take the time to prepare. Be sure you own a decent business suit. Wash. Get a "leather lunch box," or briefcase. Have business cards and stationery printed. (Never crrespond with a client on anything but your professionally printed heavyweight letterhead. And use your typewriter, or steal a friend's). Remember, you can never be too formal or polite. It is hard to take offense to proper business etiquette, although it is quite easy to have the receptionist eject a bum from your outer office or sales floor.

Here's another hint for anybody who runs any kind of business: buy a stack of classy thank you cards and use them. Thank people to pieces. Thank the creep who tells you to get lost—he may have just had a bad day, or maybe he is financially tapped out. Business people tend to congregate in flocks, and you may get a very helpful referral from an associate of the guy you thanked so professionally. I have had that happen; and nobody I know of thinks I'm a weirdo or a geek because I sent them a note of thanks. In fact, most people rarely see anything but bills and junk mail at work. At least you will stand out and be remembered. Thank the receptionist. It will probably be a first for her, and that can sure grease your way in next time. I did this once with the secretary to a sourpuss who wouldn't even take my telephone calls. The young lady was so impressed that she clued me in: "Call him early in the morning—he gets in at eight o'clock and I start at 8:30, so for a half-hour he answers the phones himself!" Can you guess whether I finally spoke with the old crab?

Speaking of thanks, I would like to extend my sincerest appreciation to all the readers who have put in a good word with my editors regarding this column, and special gratitude to all of you who took the time to send a note (so few people bother to write letters nowadays). I will try to answer any mail that comes my way, and I still hope to do a piece where I can respond to your specific questions, comments, and suggestions. No one person can know everything about producing commercials, so share ideas and success stories if you have them. By the way, I'm planning to start fooling with MIDI technology in broadcast applications, and I would like to hear of anyone's related experience in clear, simple, moron-level language.

And now I would like to say that if you have been enjoying *Ad Ventures* and you want to learn more about using your recording studio to produce radio commercials for fun and profit, buy my book, *Using Your Recording Studio To Produce Radio Commercials For Fun And Profit.* I'd like to say that, but I can't because I don't have a book out (yet). So you'll have to wait two whole months to hear from me again. And don't bother to call, because I'll be in Rio, second blanket from the right, just in front of the CinZano concession, and downwind from two mostly naked Air France stewardesses who are simply dripping glistening, aromatic cocoa butter.

Or I'll be mowing the lawn out back, just downwind from the gas can and fertilizer in the shed. ■

Sammy Caine

Tiki Recording Studios

Profiling Long Island's Tiki Recording Studios

Tiki Recording Studios (featured on this month's cover) is located on the affluent North Shore of Long Island, about 30 miles east of New York City. It is an area generally known for its great shopp-ping, tree-lined streets with expensive and expansive homes, and beautiful landscapes, but not for its first class, state of-the-art 24-track recording studios.

So why build one there instead of in New York City which is so close? "We live right around here," is one of the reasons cited by Tiki's co-owner Fred Guarino Sr. The studio is located in, what appears from the outside, to be a large private home. In fact, "It is a residential building zoned for commercial use that we purchased to use for the studio," says co-owner Fred Guarino Jr.

The house itself—all four stories—is used exclusively by the studio. Upon entrance, there is a nicely furnished waiting and reception area. The ground floor is also where one would find Tiki's well equipped Studio A with its

1. The outside of the Tiki Recording Studio building, once a private home.

2. Main room of Studio A. The booth in the back is a vocal booth, and the door on the right leads to the guitar room.

3. Guitar room with synthesizers and Marshall guitar amplifiers.

4. Tiki's waiting room and reception area.

Trident TSM 40 x 32 console and Studer tape machines. The studio itself is actually sunk about four feet below the ground floor level, so there are a few steps that lead down into the studio.

On the second floor are a few offices, a tape library, and a kitchen that is available to anyone using the studio. Up one more flight is Tiki's Studio B which is a small 4-track audio/video production room. The building's basement is used for storage.

All of this is fine and well, but does Tiki draw clients away from the first-class New York City studios? According to the co-owners, in a word, "Yes. We represent a nice reduction in a recording budget," adds Fred Sr. There is also a lot of parking right in front of the studio, something none of the NYC studios can claim.

Although they don't get as much jingle and advertising work as they might in New York City, and most of Tiki's clients are local talent, they also have quite a few national clients. Some national producers and artists that share Tiki's Studio A with local artists are clients such as Rob Freeman, Brian Setzer, Duke Jupiter, Nils Lofgren, and Melanie. Fred Jr. explains, "As we start working with certain producers we will get more and more clients. They like the fact that there is no other work going on when they're here." In fact, the clients get the run of the whole place, kitchen included, so it's almost like a home away from home. Fred Sr. adds, "If I can get them to visit, they usually book time."

Since most music recording is done in the afternoons, evening, and into the night, the most imperative thing to do is find work for the mornings. So the studio also has a lot of corporate accounts to do film strips and soundtracks for instructional films. Most of this work is done by the staff at the studio, so it can be done at any convenient time. On the average, Tiki is booked about eighty to ninety hours a week, with most of the work being done in Studio A.

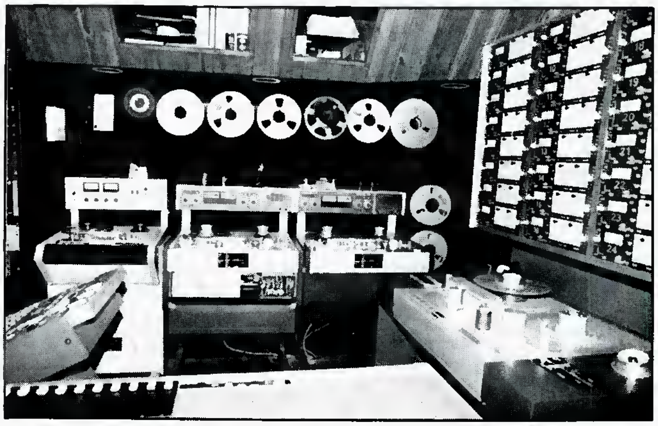
LAYOUT

Studio A actually encompasses four rooms. The control room is literally built around the Trident, with dimensions of 21 feet x 16 feet. The control room glass overlooks the adjacent main studio that has dimensions of 40 feet x 25 feet with a thirteen foot high ceiling. There are also small booths for vocals, drums and guitar recording and overdubs. Since these rooms are not in direct view of the control room, closed circuit video monitors are used.

RECENT RENOVATIONS

To provide a better control room for their clients, the Guarinos recently performed some redesigning of the room. "We repositioned the monitors and put in a lot of acoustical treatment devices and equalizers," says Fred Jr. The monitors are now very close to the console so, according to Fred, "you don't have to crank them, and the producers and engineers like that." It also provides less room coloration and better imaging, both of which will aid in a better final mix.

5. Tiki's Studio A control room featuring Trident console, UREI 813As and Yamaha NS-10M monitor speakers.



6. Tape bay in Studio A featuring Studer, Otari, and 3M tape machines.

Equipment List

CONSOLE
Trident TSM 40 x 32 console

TAPE MACHINES
Studer A80 MkIII 24-track
Studer A80 MkII 1/2-inch 2-track
Studer A80 MkII 1/4-inch 2-track
Otari MTR-10 1/4-inch 2-track
3M M79 2-track
TEAC 8-track
TEAC 4-track
TEAC 2-track (3)
Onkyo cassette decks (4)
Sony cassette decks (3)

MONITOR SPEAKERS
UREI 813As
Yamaha NS-10Ms
JBL 4311s
JBL 216s
Auratone 5Cs
Tannoy NFM-8s

MONITOR AMPLIFIERS
UREI 6300
Yamaha P-2100
Yamaha 2050
Crown DC-150A (4)
McMartin MS752

OUTBOARD EQUIPMENT
Lexicon 200 digital reverb
Ecoplate II
Yamaha REV-7 digital reverb
Lexicon Prime Time II w/MEO digital delay
Lexicon PCM42 w/MEO digital delay (2)
Eventide Harmonizers (2)
DeltaLab 1024 Effectron digital delay (3)
Korg SD-2000 sampling digital delay
Kepex II (4)
Kepex I (7)
Gain Brain II (2)
UREI 1176 LN (2)
UREI LA3A compressor/limiters (2)
dbx 160 compressor/limiter (2)
Trident stereo compressor/limiter
Ashly SC-50 compressor/limiters (2)
Symetrix CL-100 compressor (2)
Aphex Aural Exciter (2)
Eventide 201 flanger
Orban 516EC de-esser
Orban 622 equalizer
Isomix HP-4 4-channel headphone systems
Ashly SC-66A equalizer
Roland 830 phaser

MICROPHONES
Neumann
AKG
Milab
Sennheiser
Beyer
Electro-Voice
Shure
Audio-Technica
Sony

INSTRUMENTS
Yamaha C-5 grand piano
Slingerland drums
LinnDrum machine with extra chips
Kaypro PC 20 meg computer (IBM compatible)
Korg Poly 6
Hammond organ with Leslie
Roland, Marshall, Ampeg, and Fender amplifiers

EQUIPMENT

The custom-made Trident TSM 40 x 32 console is the center point of the control room. Just above its meter bridge are a pair of UREI 813As and a pair of Yamaha NS-10s that look like guppies next to whales. To the left of the console is a Studer A80 MkIII 24-track tape machine. Opposite the console is a bank of 2-track machines—a Studer MkII, an Otari MTR-10, and a 3M M79. There are also two racks of outboard processing gear, and several Teacs—one 8-track, one 4-track, and three 2-track copy machines.

7. Studio A control room featuring two full racks of outboard gear.

Currently, Fred Jr. and Fred Sr. are considering the purchase of a large amount of MIDI equipment to make a "MIDI Room." They both find MIDI to be fascinating and are trying to decide on a computer system for the basis of the room—the IBM PC or the Apple Macintosh.

When asked if there is a digital future for Tiki, Fred Jr. says, "We cannot afford a digital recording system at our current studio rates, and we cannot raise our rates and expect to keep our clients."

But the owners don't think this will hurt their business in the future, nor do they feel home recording affects professional studios. As Fred Jr. states, "You still need a professional room to work in. Our control room is not merely a living room with speakers, it is a fine tuned machine." ■

Len Feldman

db TEST

Shure Brothers HTS5000 Surround Audio Processor

GENERAL INFORMATION

THE SHURE HTS5000 Surround Audio processor was designed to provide a theater-sound environment in the home, office, or club . The HTS stands for Home Theater Sound, and this unit is specifically intended for decoding the many video discs and video cassettes and even the sound tracks of movies broadcast over TV stations, if those motion pictures were originally encoded with Dolby stereo sound. The HTS5000 provides up to six channels of audio output: left, center, right, two surround channels, and a sub-woofer output. Shure has developed a kind of decoding logic which they call Acra-Vector which improves the directional characteristics of the surround channels beyond that of a simple "matrix" decoder.

A true digital time delay system provides cleaner surround channel outputs than has been possible up to now using so-called "bucket brigade" or CCD analog devices. A basic HTS Surround Sound installation would employ four loudspeakers and, of course, two separate stereo amplifiers. A more complete system would use six loudspeakers and

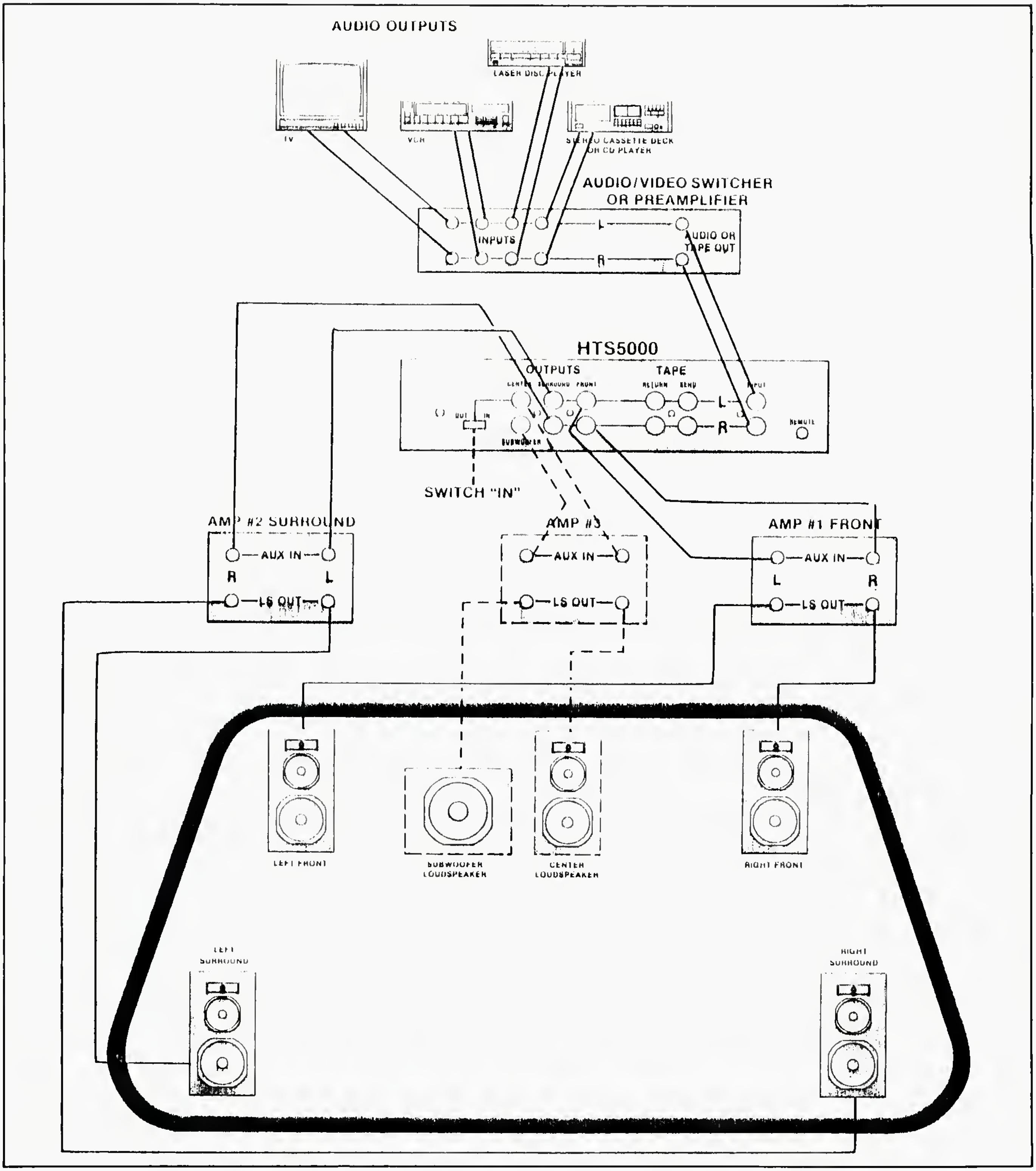


Figure 1A. Basic and complete systems using an audio/video switcher or preamplifier.

three stereo amplifiers to reproduce the full "theater experience," as illustrated in *Figure 1A* (using an audio/video switcher or preamplifier and three stereo power amplifiers) or *Figure 1B* (in which a stereo receiver and two power amplifiers are used).

When connected in a system as shown, the Shure Acra-Vector circuits enhance the directional accuracy of the reproduced sound field. In the front, this creates a stereo panorama with highly realistic motion effects from moving sources. The "surround sound" channel signals are digitally time delayed by amounts chosen by the user, depending upon the room in which the system is installed. The delayed signals are further processed to enhance the feeling of spaciousness by circuits which Shure calls their Acoustic Space Generator. In the basic system, dialogue originates from a phantom center channel, while in the more complete system, a center-front speaker delivers the dialogue. The center-front speaker helps to keep the acoustic image stably centered no matter where the listener sits within the listening room.

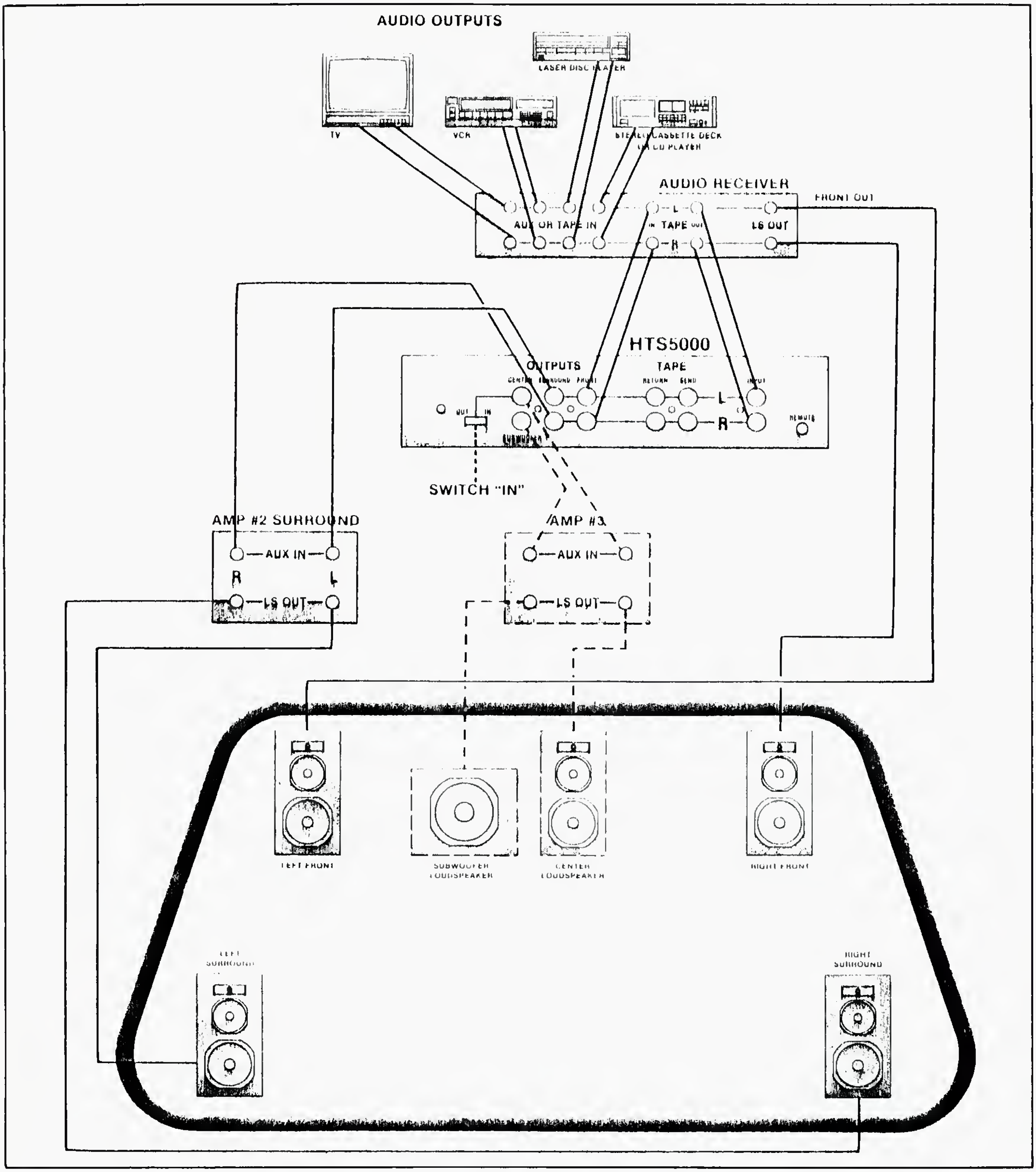


Figure 1B. Basic and complete systems using an existing receiver and stereo loudspeakers.

Finally, in the complete six-speaker system, low frequency information below 80 Hz is assigned to a sub-woofer channel. Use of a sub-woofer reinforces the theater illusion, since the film industry often uses very low frequency audio signals to create realism of mood and place depicted in the film.

MATRIX ENCODING AND DECODING

Those readers who are relative newcomers to the world of audio may be wondering how it is possible to achieve "surround sound" and directional characteristics that move all around the listener from only two actual audio tracks. For the benefit of those readers (and for older readers who have perhaps forgotten all about 4-channel or "quadraphonic" sound as practiced in the early 1970s), a quick review of the basic of 4-2-4 Matrix encoding and decoding may be helpful. Back in those 4-channel days, speakers were arranged as shown in *Figure 2*, with the mutual separation of each speaker pair shown between pairs of arrows. These numbers are obtained with the original Peter Scheiber matrix equations as well as with the so-called Sansui QS matrix and several other symmetrical matrix systems. All of these systems use the same amplitude ratios of L_T and R_T (L_T and R_T are the two predominantly left and predominantly right transmission channels) to represent sound source directions corresponding to each of the four speaker locations. Differing relative phases between L_T and R_T for the various matrices' directional encodings result in differing directional encoding and decoding capabilities.

The CBS SQ matrix, illustrated in *Figure 3*, trades decreased front-to-back separation for full left-front to right-front and left-back to right-back separation. The front stage is essentially identical to conventional two-speaker stereo, and has full stage width, but center image

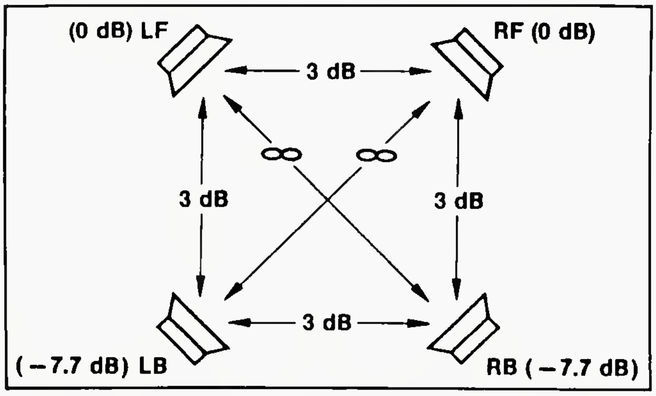


Figure 2. Symmetrical interspeaker leakages of several quadrophonic matrices. (Response to center front dialog in parentheses.

stability is highly dependent on left-right listener positioning.

In the Dolby Surround matrix, illustrated in *Figure 4*, the surround speaker array signal S' is represented by a single speaker. (L',C',R' and S' are the decoded signals before directional enhancement.) Unlike earlier quadraphonic systems, the speaker array is not symmetrical from front to back. Most of the matrix's directional encoding space is devoted to the front soundstage. The full separation of left and right loudspeakers enables a wide soundstage narrowed only by about twenty-five percent from the full left-right speaker, spacing by the presence of the center speaker separated 3 dB from the left and right speakers. In a home installation this can be compensated for by somewhat wider speaker spacing.

The decoding matrix for the Dolby surround sound system (before extra directional enhancement provided by Shure's patent-pending Acra-Vector system) is defined by the following four equations:

$$L'=L_T$$
$$R'=R_T$$
$$C'=0.707(L_T + R_T)$$
$$S'=0.707(L_T - R_T)$$

The encoding equations used for Dolby surround sound are not quite complementary to the decoding equations, in that they introduce a relative phase shift between L_T and R_T (other than zero percent or 180 percent) for signals "panned" between surround and any front direction input. The encoding equations are:

$$L_T = L + 0.707\ (C\ \text{-}jS)$$
$$R_T = R + 0.707\ (C +jS)$$

L, R, C, and S are the input signals to the encoder intended to come predominantly from their respective speakers after decoding. The -j coefficient causes the S signal to be mixed into L_T with a 90-degree phase lag relative to the front signals and the +j with a 90-degree phase lead into R_T relative to the front signals. C is mixed into the L_T and R_T in-phase at equal 3 dB attenuated levels while S is mixed into L_T and R_T in opposite polarity (180 degrees out-of-phase at equal 3 dB attenuated levels.

As *Figure 4* shows, quite a bit of cross-talk remains. The means to add directional enhancement have evolved and been improved since the early days of quadraphonics. The Shure Acra-Vector directional enhancement circuitry has been designed to enhance predominant sounds' localizations for any decoded direction, including panned sources between left and center and between right and center, while providing power compensation for those sources appropri-

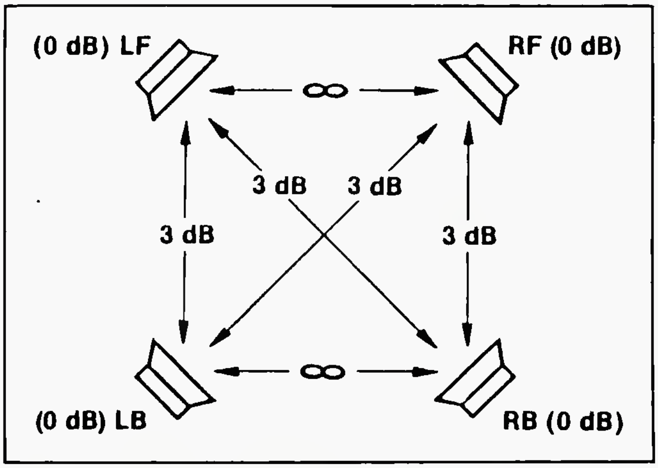


Figure 3. Interspeaker leakages of the SQ matrix. (Response to center front dialog in parentheses.)

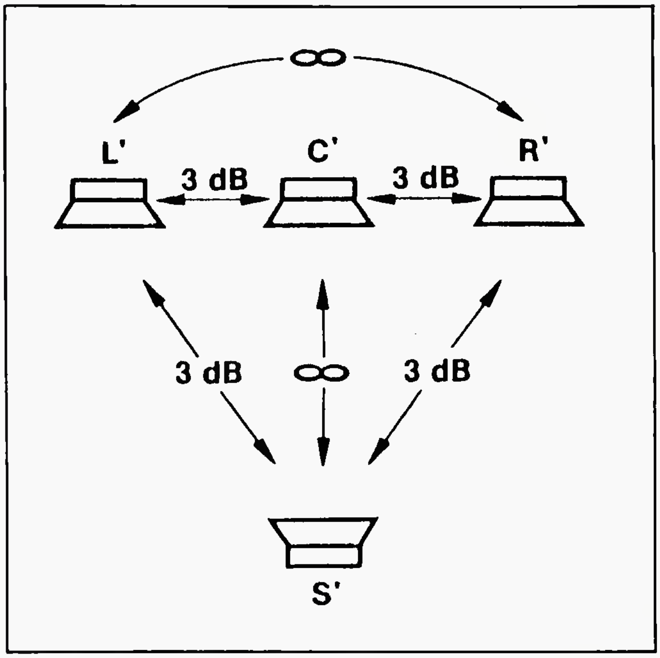


Figure 4. Interspeaker leakages of the Dolby Surround Matrix before directional enhancement.

ate for encodings made from the discrete original master tapes. Furthermore, if a user elects not to use a center speaker (initially, or ever), a switchable option is provided to vary the characteristics of the directional enhancement circuitry so that the center sounds are not enhanced to a non-present speaker. A block diagram of the Shure HTS5000, as con*Figured for the Dolby Surround mode, is shown in Figure 5.*

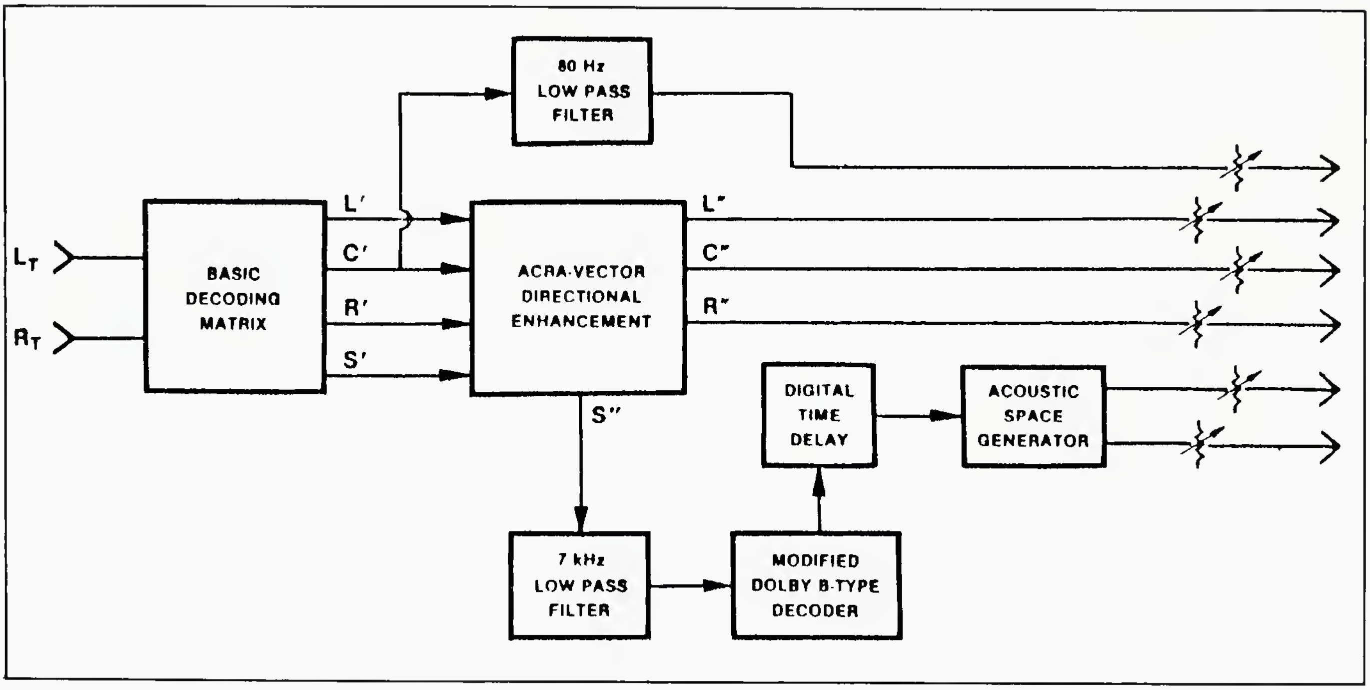


Figure 5. Block diagram of the HTS 5000 in the Dolby Surround mode.

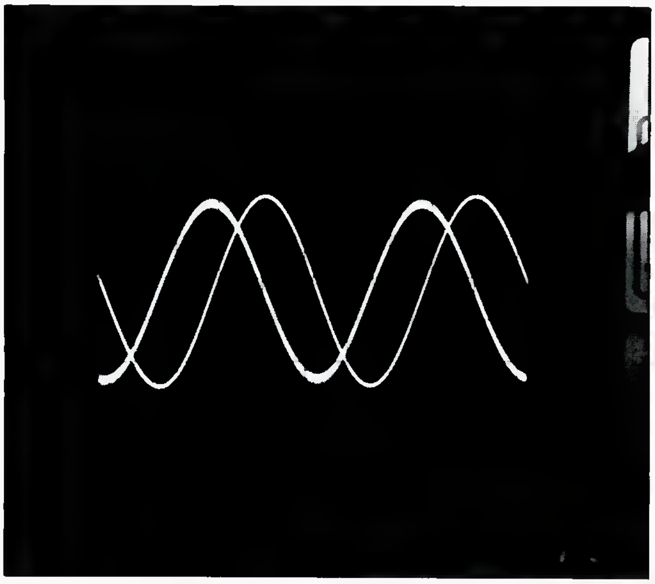
CONTROL LAYOUT

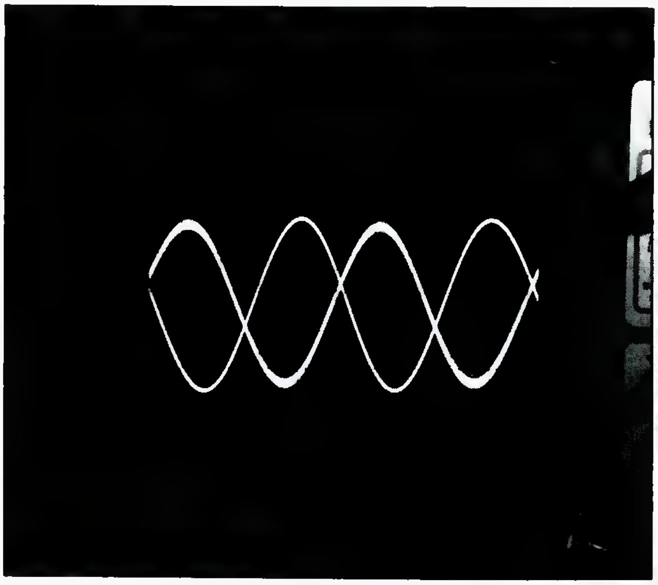
The HTS5000 front panel controls and displays include pushbutton selectors of operating modes: Dolby Surround, synthesized-surround stereo, or mono and, by means of a defeat button, pure stereo or mono. There is also a pushbutton choice of the audio source from either Left and Right inputs or from Tape Monitor Return inputs. Rotary knobs on the front panel adjust the following: input level, with a five LED level display; input balance, with an LED for determining the correct balance; digital delay time best suited to the particular listening room; and volume and surround levels. An overriding wired hand-held Remote Control allows you to adjust surround and overall volume levels from the listening position. The Remote also permits muting of the audio inputs. A four-channel graphic display on the front panel provides a visual diagram type of indication of the surround characteristics of the input signals.

The HTS5000 rear panel includes left and right audio inputs; outputs for front left and right and surround left and right channels; a separately switched front-center output jack and a sub-woofer output jack. Tape monitor Send outputs and Return inputs for left and right channels are also found here. These audio input and output jacks are all phono types. In addition, front and rear panel eighth-inch diameter phone jacks accommodate the wired remote control attachment.

LAB MEASUREMENTS

A complete table of VITAL STATISTICS, covering the manufacturer's published performance specifications and our own lab test results will be found at the conclusion of





Figures 6A and 6B. Input and surround channel output signals are compared with time delay set to minimum (A) and to maximum (B).

this report. As you might have expected from a firm with the reputation of Shure Brothers, all of the basic response and distortion measurements were either met or exceeded. As for signal-to-noise measurements, in the few instances where our sample was off by a couple of dB, the differences were so minor as to be considered negligible.

Figures 6A and *6B* illustrate the action of the variable digital time delay circuitry which forms a part of this system. In *Figure 6A* we superimposed the input and surround output signals in a dual-trace 'scope presentation, with the time delay control set for minimum (sixteen milliseconds). Horizontal sweep rate was set for twenty milliseconds per division and, as you can see, the space between the input and output traces is just short of one horizontal division. Turning the time delay control fully clockwise (thirty-six millisecond delay), we took another photo (*Figure 6B*) and this time, the distance between the two traces is closer to two full divisions, as it should be.

By far the greatest amount of time spent with the HTS5000 was not on the lab bench, but rather in the listening room. The kind of listening experience that the HTS5000 provides cannot be quantified by numerical data; it has to be heard to be fully appreciated.

COMMENTS

I have heard many surround sound processors since I've been involved with audio. If you count the four-channel systems that were around in the 1970s, I've probably tested more multiple-channel home audio systems than anyone around. I make that statement because, for better or worse, I was even responsible for one of the early matrix 4-channel systems that was sold in the early 1970s (The Electro-Voice Stereo-4 system was what it was called), so you can well imagine that I checked out most of the competition. More recently, I've had occasion to check out some of the "surround sound" units that are intended primarily for use with video systems. I can say without hesitation that the Shure HTS5000 beats any of the analog-based surround sound decoders or enhancers. The sound delivered for the surround channels was cleaner, and better defined than any I have heard from earlier units. I would mention, however, that other firms have begun to employ digital technology for surround sound and ambience enhancement in home audio systems. Among the most recently announced units of this type are a Model SDP505 from Sony and a DSP-1 from Yamaha, both of which also employ digital time delay and Dolby decoding as well as other sound processing functions. Since I have not had an opportunity to test either of these units as yet, the Shure HTS5000 stands by itself as a surround sound processor that recreates the excitement of a movie theater experience and does much more besides.

Once the system is set up around the HTS5000 (and I strongly Comments recommend the six-channel approach, if budget permits), you will find that adjustment of the system for optimum effects is relatively easy, especially if you allow the excellently written owner's manual prepared by Shure to guide you, step by step, towards optimum utilization of their unit. I can't tell you how many thousands of dollars are spent to equip motion picture theaters for proper Dolby Stereo motion picture presentations, but I will wager that even if you have to go out and buy one or two extra stereo amplifiers to use with the Shure HTS5000, for home or studio Dolby Surround sound decoding you'll be way ahead both financially—and sonically!

VITAL STATISTICS SURROUND AUDIO PROCESSOR

MAKE & MODEL: Shure HTS5000

SPECIFICATION	MFR'S CLAIM	db MEASURED
Frequency Response		
Front (Left,Center,Right)	20Hz-20 kHz,+/- .5 dB	Confirmed
Sub-woofer	-3 dB at 80 Hz	Confirmed
Surround (Left, Right)	50 Hz-7 kHz, -3 dB	40 Hz-7.5 kHz
Input Sensitivity		
Input Control at Max.	0.18 V	0.185 V
Input Control at Min.	1.8 V	1.7 V
Input Clipping Level	2.8 V	3.0 V
Output Level Clipping	4.0 V	4.2 V
Input Balance Control Range	+/-9 dB	+9, -10 db.
Output Level Adjust. Range	20 dB	Confirmed
THD at 1 kHz, 1 V output		
Front (Left,Center,Right)0.1%	0.05%	
Surround (Left, Right)	0.3%	0.15%
S/N, A-weighted		
Front (Left,Center, Right)		
Volume at Center	90 dBV	88 dBV
Volume at	80 dBV	83 dBV
Surround (Left, Right)		
Vol. & Surround Ct'l.Ctr'd	.85 dBV	Confirmed
" " " Max.	68 dBV	70 dBV
Surround Time Delay Range	16 to 36 msec.	Confirmed
Power Requirements	120 VAC, 60 Hz, 36 W	Confirmed
Operating Temperature Range	-29 to 57o C	
Dimensions (H x W x D, in.)	2-3/8x16-13/16x15-1/32	
Net Weight		
HTS5000	9 lbs. 13 oz.	
Remote Control	3 oz.	
Retail Price	$599.00	

Neil A. Muncy

Applying The Reflection Free Zone RFZ®R Concept in Control Room Design

See what's to be learned about the RFZ concept!

A WELL established principle of acoustical physics is that sound energy will reflect from any rigid surface which is largely relative to the wavelength(s) of the impinging sound. In an enclosed space, the energy reflected by the wall, ceiling, and/or floor surfaces immediately adjacent to a loudspeaker will recombine with the direct sound from the loudspeaker in either a constructive or destructive manner, depending on the difference in travel paths and the frequencies involved. This recombination process can result in a radical distortion of

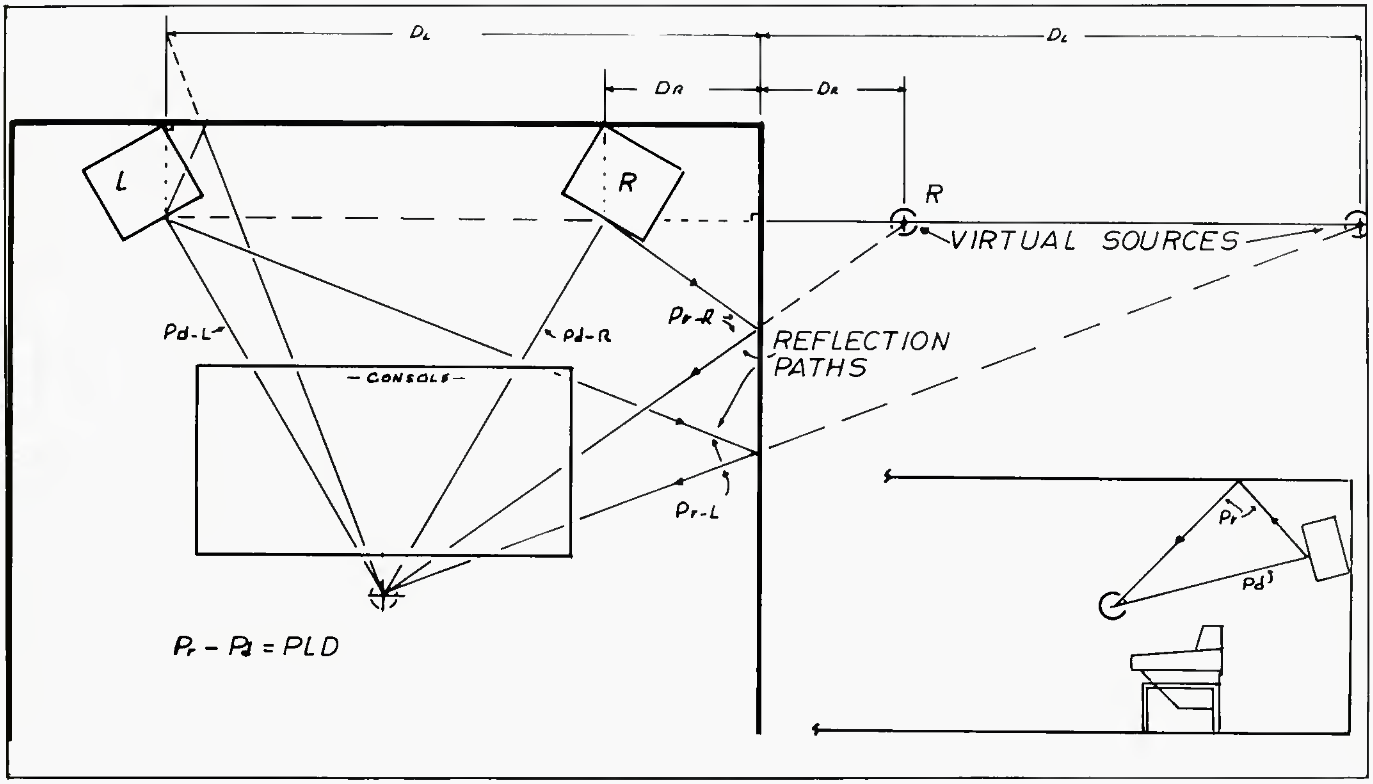


Figure 1. The floor plan and elevation of the front of a Non-RFZTMR control room.

Neil A. Muncy is the president of Neil Muncy Associates, LTD., consultants in electroacoustic systems.

the inherent frequency response of the loudspeakers, and is known as comb filtering in reference to the shape of the resulting response curve which may contain peaks and dips of more than 20 dB in typical situations! In attempting to deal with this phenomenon, it is absolutely essential to realize that comb filtering is the result of an acoustical process which is occurring in the air between the speakers and the listener!

In a typical home entertainment situation, comb filtering is always present and is usually not only tolerated, but in most cases completely unnoticed by the average listener, due at least in part to the fact that there is no way to compare the reproduced sound with the original. Ignorance can be bliss!

In a control room, however, both conscious and unconscious decisions regarding level, balance, and timbral aspects of the sound are continuously being made by the engineer in response to what he or she hears. These decisions are translated into control operations including balance, equalization, and reverberation changes, etc., which are then passed along to the recorder(s). If the frequency response of the sound arriving at the engineer's ears is significantly different from that being sent to the recorder, the decisions made by the engineer may result in an "equalized," "balanced," and "reverberated" recording which will only be reproducible with any degree of satisfaction in the control room itself.

In an attempt to eliminate this problem, it has been common practice to "equalize" the monitor system to obtain either a "flat" reading or some other curve as displayed on a Real Time Analyzer (RTA) or other measurement device. While making the owners of the studio quite "happy" with their nice, even RTA display, the results of subsequent recording efforts may not be nearly as satisfactory. In spite of the equalization applied to produce a flat reading on the measuring equipment (which does NOT operate the way our ear-brain system does!), the comb filtering process is still happening out in the air in front of the speakers, and still influencing the decisions of the operator. The bottom line is that one cannot employ an electrical means to "fix" this type of acoustical problem.

If the loudspeakers to be installed are capable of providing essentially flat on and off-axis response under anechoic conditions, and if it is desired to approximate this response along the principal console operating area, then the geometry and surface treatment of the room boundaries between the loudspeakers and the engineer must be arranged so that reflections from these boundaries are reduced to negligible proportions. In other words, what is needed is a means of placing the engineer in a "Reflection Free Zone."

The Reflection Free Zone®R concept was presented by Dr. Peter D'Antonio at the 76th Convention of the A.E.S. in October, 1984. (1) Dr. D'Antonio defines a Reflection Free Zone as an area in a control room "where the predominant (acoustical) energy is from the monitor speakers." Ideally, this area should include that principal seating position of both the engineer and the producer, and should be wide enough to permit a fair degree to freedom of lateral movement along the console.

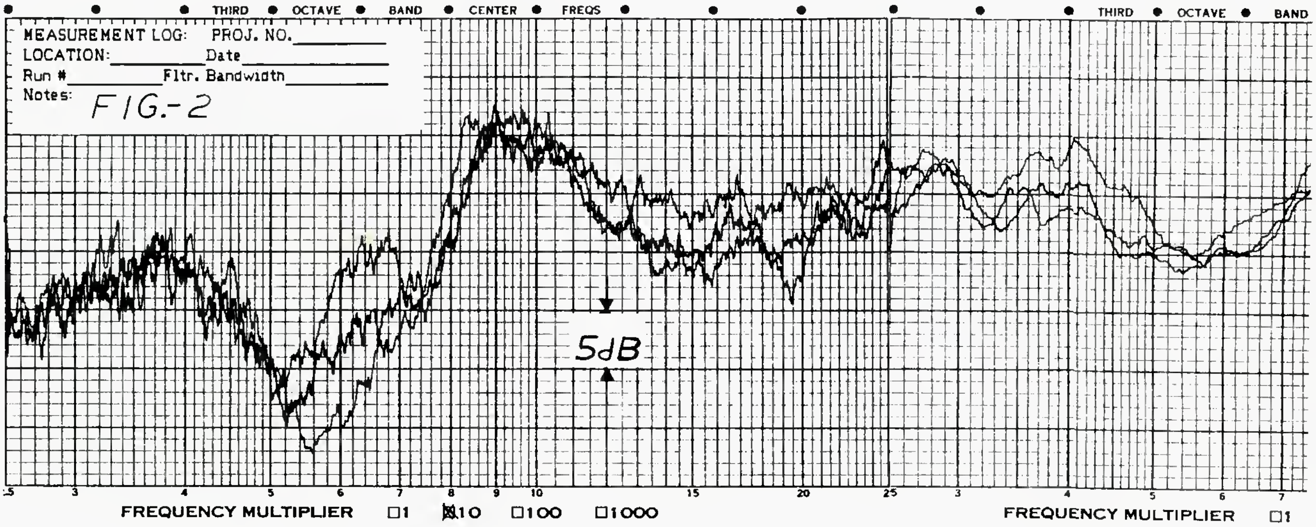


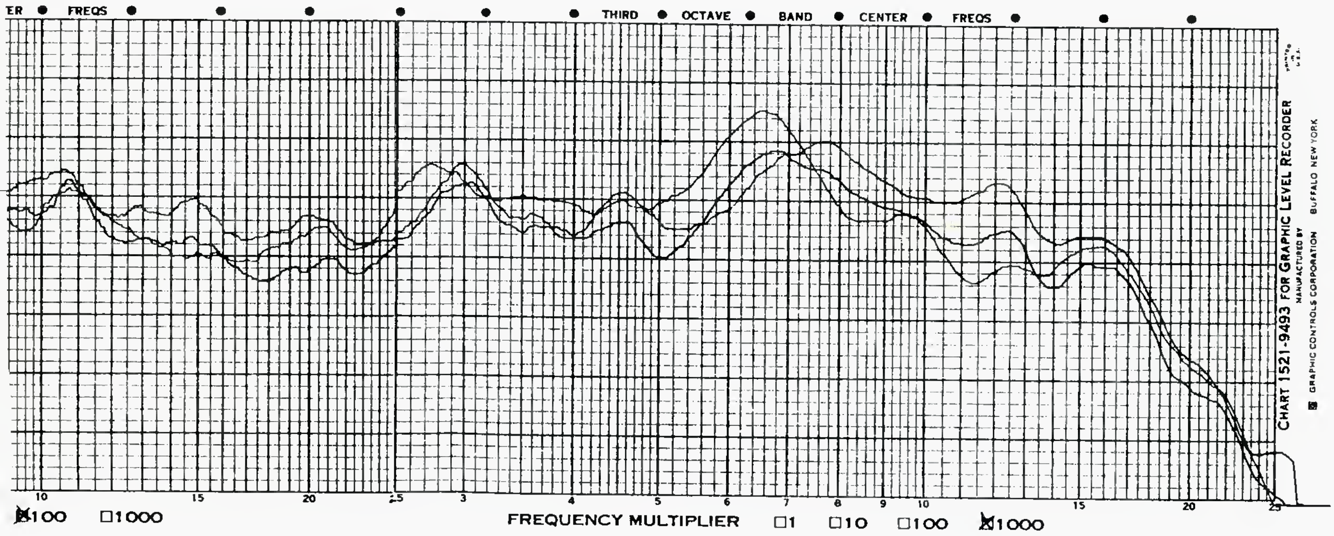
Figure 2. The monitor response measured in a control room in which reflected energy is returned in the

In building a control room, there are two ways to minimize boundary reflections. One way is to make use of the Inverse Square Law and simply bring the loudspeakers close to the operator—the "Near-Field" approach. While undoubtedly reducing the magnitude of the reflections from the room boundaries, this scheme usually results in a significant reflection from the console surface—in effect, trading one kind of problem for another. An alternative approach involves building the loudspeakers into the intersections of the front room boundaries so that reflections are precluded except at very short wavelengths.

To illustrate the value of this concept, first consider the floor plan and elevation of the front of a Non-RFZ®R control room as shown in *Figure 1.* In this layout, which is typical of countless installations, the loudspeakers are located some distance from the adjacent front, side, and ceiling boundaries. By using a simple architectural process called "Ray-Tracing," the influence of a reflection from any one surface can be predicted by locating the "virtual source" of reflected energy along a line perpendicular to the surface. The Direct path length (Pd) is subtracted from the Reflection path length (Pr), yielding the Path Length Difference, (or PLD). When the PLD is substituted for "C" in the wavelength formula, V equals FC, where V equals the speed of sound (1130 Ft/sec), C equals the wavelength (Feet) and F equals the corresponding frequency, a set of interference frequencies can be calculated. For PLD's equals 1/2, 1 1/2, 2 1/2, 3

1/2 etc. wavelengths, the reflected energy will produce a dip in the response. For PLD's equals 1, 2, 3, 4, etc. wavelengths, peaks will be present. These frequencies define the centers of the peaks and dips which will be observed in the response curve.

Multiple reflecting surfaces will produce peaks and dips at frequencies directly related to their distance from the source and the observer. *Figure 2* illustrates the monitor response measured in a control room in which considerable reflected energy is returned in the same approximated time frame from both sidewalls and the ceiling. The abrupt drop in response below 100 Hz (almost 25 dB) is the result of several reflection paths of similar length "ganging up" to produce a severe response distortion. *Figure 3* illustrates the extent of the reflection coverage in a control room incorporating this type of loudspeaker placement. While there is no question that this technique is inexpensive, convenient and easy to implement, the resulting comb filtering will make the flattest loudspeaker appear to be a poor performer at best, and electrical equalization of the



same time frame from both sidewalls and ceiling.

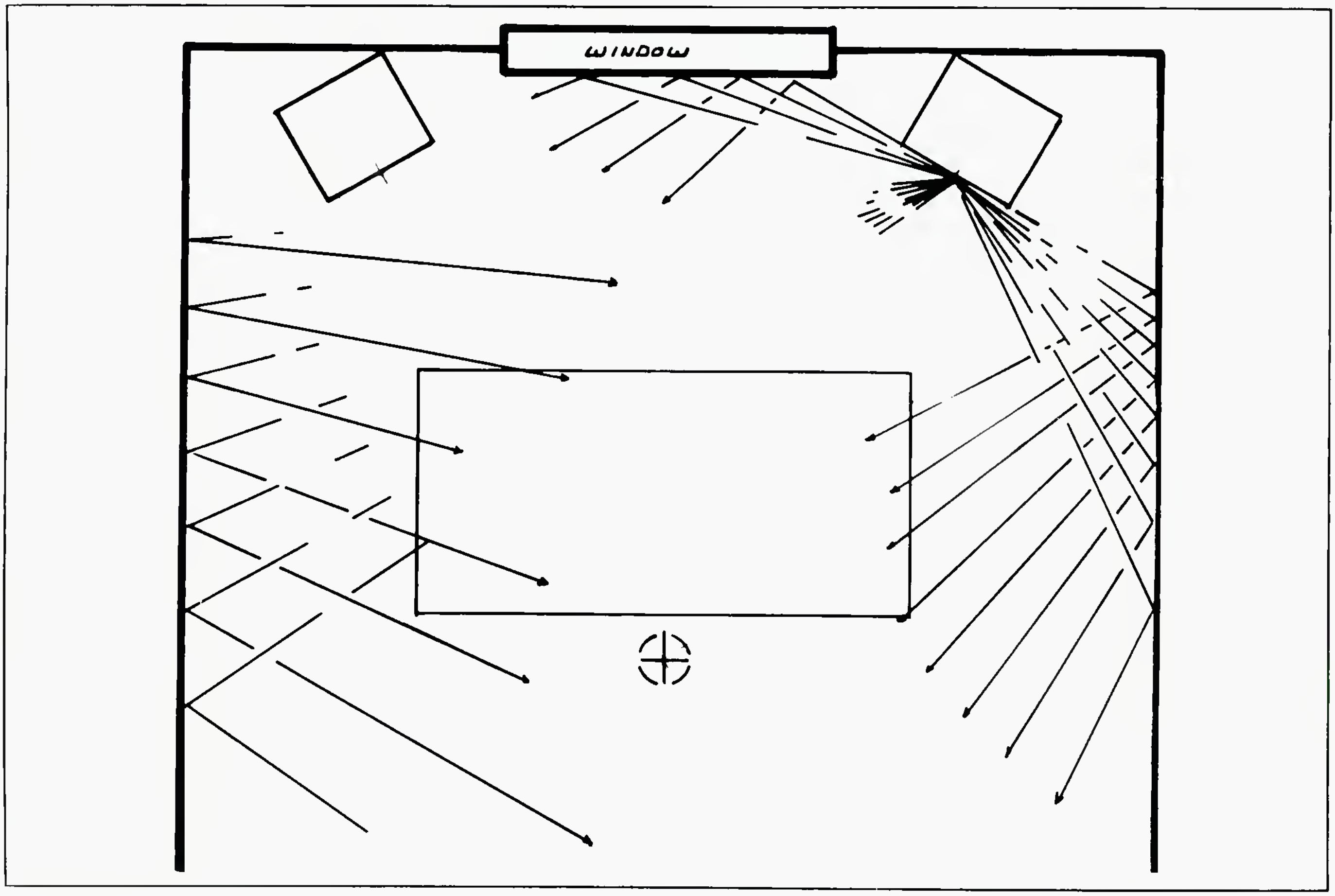


Figure 3. The extent of the reflection coverage in the control room.

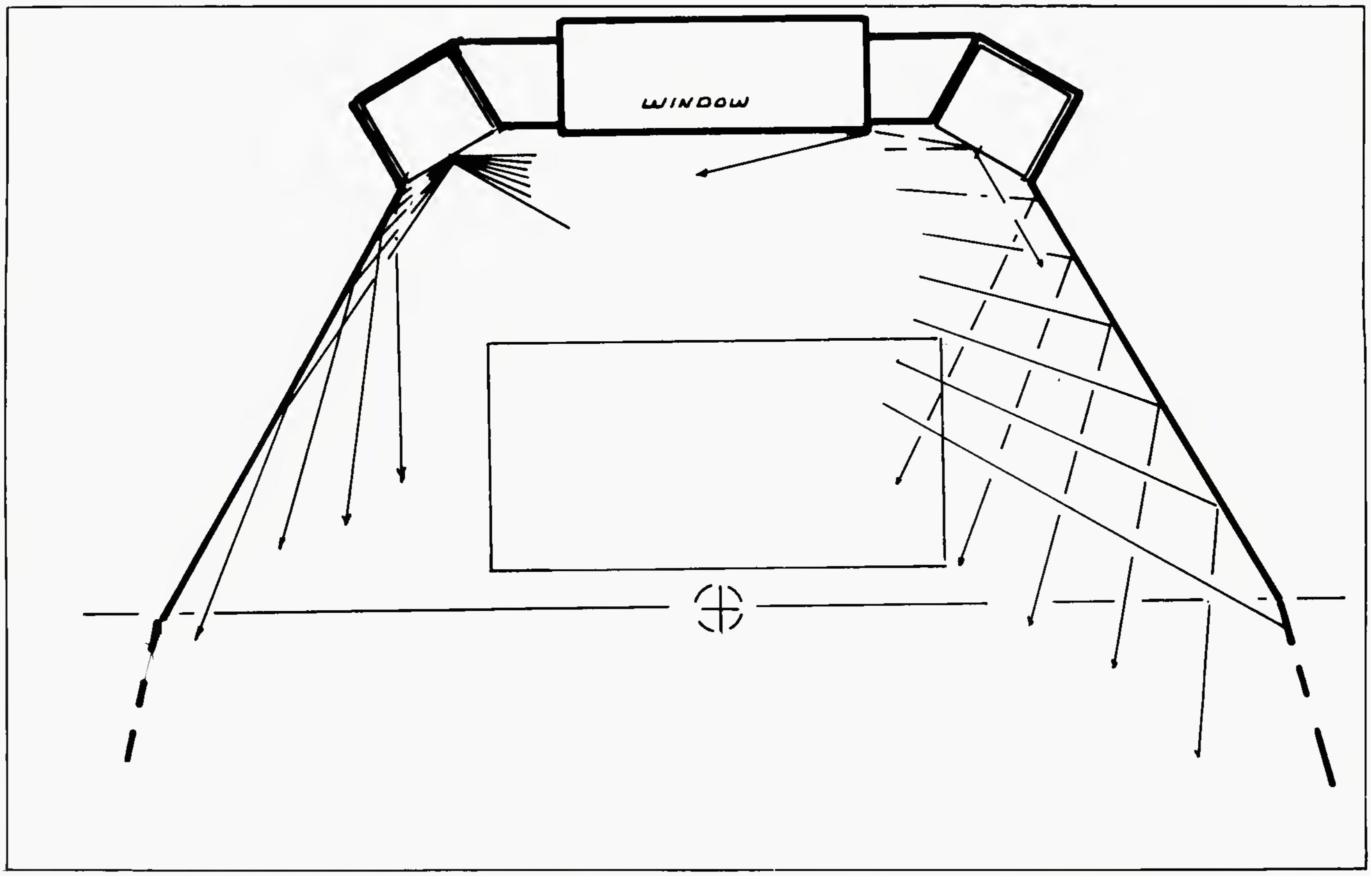


Figure 4. Splaying the walls and ceilings to produce a Reflection Free Zone around the console.

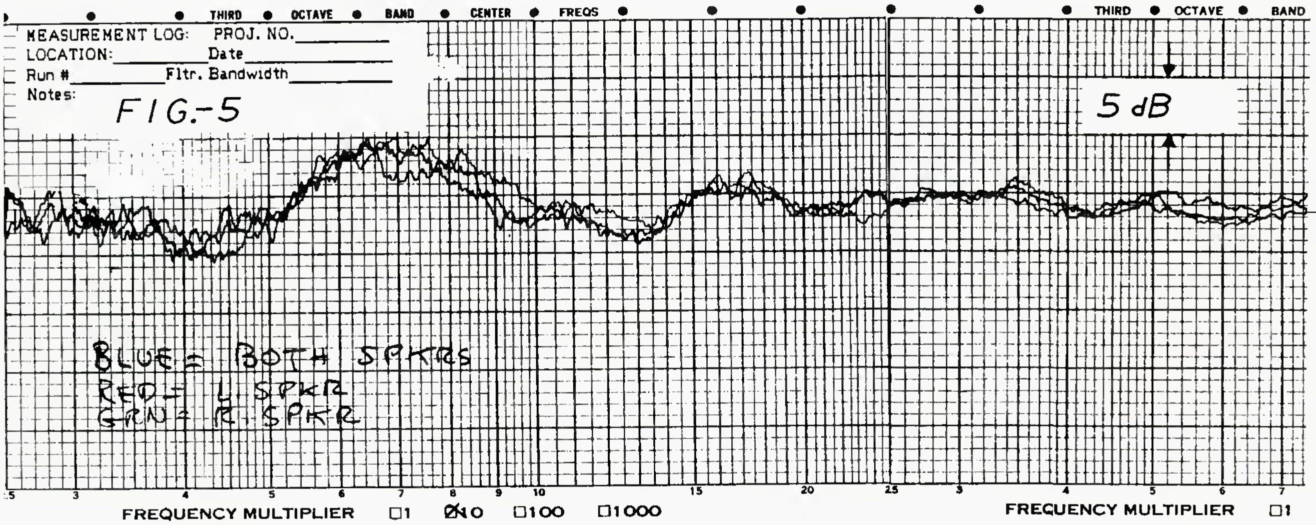


Figure 5. An example of an early control room.

system will only smother the real problem.

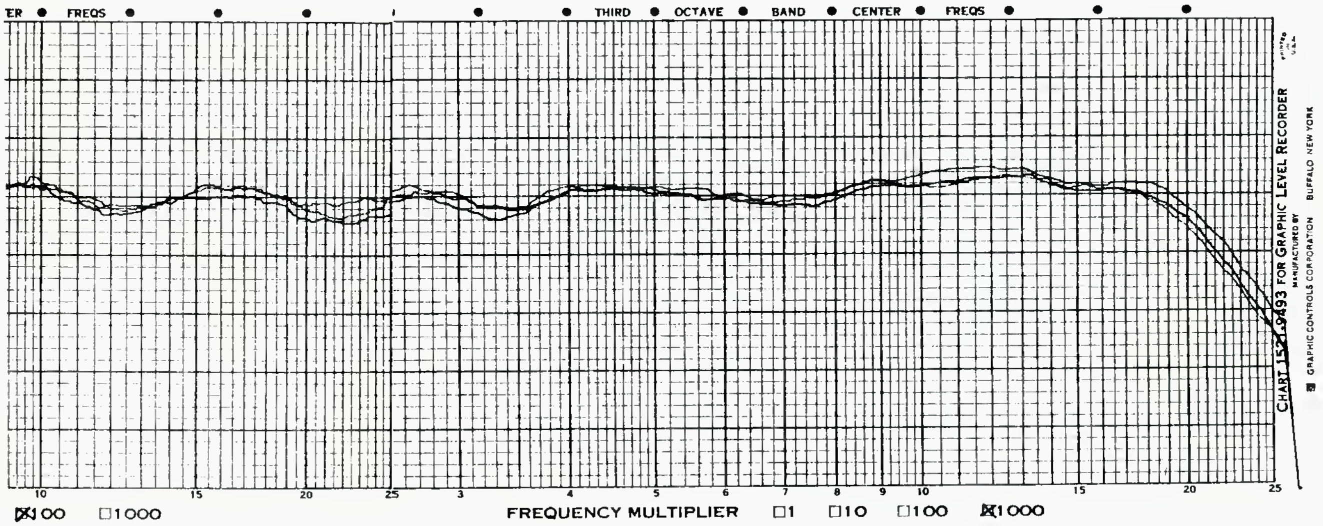
We will now rearrange the front boundaries of the room into a configuration which will produce a Refection Free Zone around the console. This is done by splaying the walls and ceiling so that the loudspeaker cabinets are literally built into the resulting trihedral corner, as shown in *Figure 4*. In mounting the loudspeakers, it should be noted that the most important driver in this scheme is the woofer. By orienting the loudspeaker cabinet so that the woofer is as close to the corner as possible, the immediately adjacent surfaces will be sufficiently close to preclude out-of-phase reflections at long wavelengths. Reflections from the other drivers at shorter wavelengths can then be minimized by the use of absorbent wall covering. A point to remember is that loudspeakers in which all drivers are mounted in a straight line along the vertical axis of the cabinet usually exhibit a broad horizontal polar pattern, and thus will make a particularly good choice, as the polar pattern will usually be wide enough to cover the operating area of even a rather large console. This driver arrangement also results in all drivers being equidistant from the front and side walls, leaving only the ceiling surface as a potential source of major reflections from the high frequency drivers.

A fringe benefit of this arrangement is the increase in bass efficiency realized by corner-mounting a loudspeaker which was designed to operate into a hemispherically shaped space. At low frequencies, the polar pattern of most loudspeakers is essentialy omnidirectional, and in a typical application the low frequency energy spreads out spherically to fill the entire space surrounding the speaker. The high frequency drivers usually develop a much narrower polar pattern than the woofer, meaning that most of the high frequency energy is concentrated in a smaller portion of the space in front of the speaker. To compensate, the efficiency of the drivers is balanced to yield on-axis response which is "flat" down to some lower cutoff frequency which is determined by the size of the woofer cone and the dimensions of the cabinet. When such a loudspeaker is mounted in a corner, the low frequency output of the speaker is concentrated into a much smaller space than originally intended, resulting in a bass buildup. This increase in efficiency can be neutralized by the application of a gentle low frequency shelving rolloff ahead of the power amplifier. For a given SPL in the room, the amplifier will then be required to deliver considerably less power at low frequencies, resulting in lower distortion and greater headroom in the monitor chain. Depending on the original design of the loudspeaker system, increases of efficiency of up to 10 dB may be realized by this technique.

At very low frquencies, the overall response of the monitor system will be governed by the size, shape, and volume of the room. As a result, it is inevitable that the low frequency response will deviate from total flatness. Smaller rooms will display a greater tendency to get "bumpy" and

measured in a continuous and very slowly swept 1/3 octave bandwidth. The microphone was carefully placed on the median plane at the crossing point of the loudspeakers at the principal mix position. The three overlaid curves represent left-only, right-only, and both speakers together with the level reduced 6 dB. There is no equalization of any kind in the system. The low frequency buildup due to corner mounting was neutralized by the use of an electronic crossover to control the level fed to the woofer, and the choice of crossover frequency. The bumpiness of the response below 200 Hz can be directly traced to the energy reflected from the rear wall. After experimenting with equalization to reduce the bumpiness, it was determined that the audible improvement with equalization did not justify the cost of the equalizers.

This brief overview of the RFZ®R technique is by no means a complete design guide, but should give the reader a general appreciation of the value of this technique. The use



will do so at a higher frequency than larger rooms. In a room constructed in the RFZ®R manner, the largest reflecting surface will usually be the rear wall. The construction of this surface could easily be the subject of another article, but suffice it to say that the rear wall must incoporate as much low frequency diffusion as possible, and then provide as much absorption as possible below the frequency where diffusion is no longer achievable.

An example of the behavior of an early RFZ®R control room providing about 350 feet of floor space is shown in *Figure 5*. This measurement was made using pink noise

of the RFZ®R technique is another tool in the evolving process of creating "acoustically neutral" control room spaces, making it increasingly possible to produce recordings with a high degree of credibility wherever people listen to recorded sound.

References: (1) D'Antonio, P. and Konnert, J., "The RFZ/RPG Approach to Control Room Monitoring," AES Preprint NO.2157 (I-6), presented at the 76th AES Convention 1984 October 8-11.

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Drew Daniels

Vented Loudspeaker Enclosure Design Made Easy

The most commonly asked questions about building enclosures.

MANY JBL users build their own loudspeaker enclosures. Their audio skills range widely from novice to expert. From the thousands of letters and calls we have received addressing the subject of loudspeaker enclosure construction, we have determined which are the most common questions and present the following questions and answers. Those listed attempt to answer as many questions as we feel are necessary to provide enough information to build an enclosure which will allow your JBL loudspeaker to operate to its potential. The questions selected here concentrate on vented "bass reflex" enclosures, since low frequency horns are fairly complex, and many good tested designs exist. Also, it is often more econonomical to buy a bass horn enclosure than to build one. Vented box enclosures are by far the most popular enclosure type. Vented boxes are finding increasing use by touring sound companies, displacing existing horn enclosure designs because of the greater low frequency power output and extended low frequency capability they offer when used in large arrays. In addition to their simple design requirements, vented loudspeaker enclosures offer flexibility of design in shape, weight and component complement, and usually produce the best results obtainable from modern loudspeaker drivers at the lowest cost.

[1] Q: What makes a good vented enclosure?

A: Basically, an enclosure serves to partition the front and rear of the driver's cone, preventing the opposing air pressure changes produced by cone motion from cancelling, and allowing the radiation of sound from the front of the driver only. In addition, vented enclosures allow the compressibility of the air inside the enclosure to work as a more active part of the "system" consisting of driver and enclosure. Beyond these two basic functions, a low frequency loudspeaker enclosure should do absolutely nothing, that is, it should add no effects of its own—no vibration, no tonality, no motion—nothing to interfere with or absorb acoustic energy produced by the driver.

[2] Q: Is it possible to get low, punchy bass from a small enclosure?

A: Yes, if the driver in the enclosure is designed for low bass operation in a small enclosure. Unfortunately, it's usually a small driver that can work properly in a small enclosure, and that dictates that lower sound levels will result from the small amount of air such a small driver can move. Larger boxes (with larger bass drivers) produce more bass, smaller boxes produce less bass. It's a fact of life, like the fact that it takes a bass viol, a tuba, longer piano strings, or very large organ pipes to produce bass energy in the air. Low bass requires that more air move, and bigger boxes contain more air that can be put to work making low bass.

[3] Q: Can I get more bass from my enclosure by installing a bigger driver?

A: A given enclosure will not automatically produce more bass when a larger driver is installed. In fact, the opposite is often the result.

[4] Q: What about putting two drivers in the enclosure to increase bass?

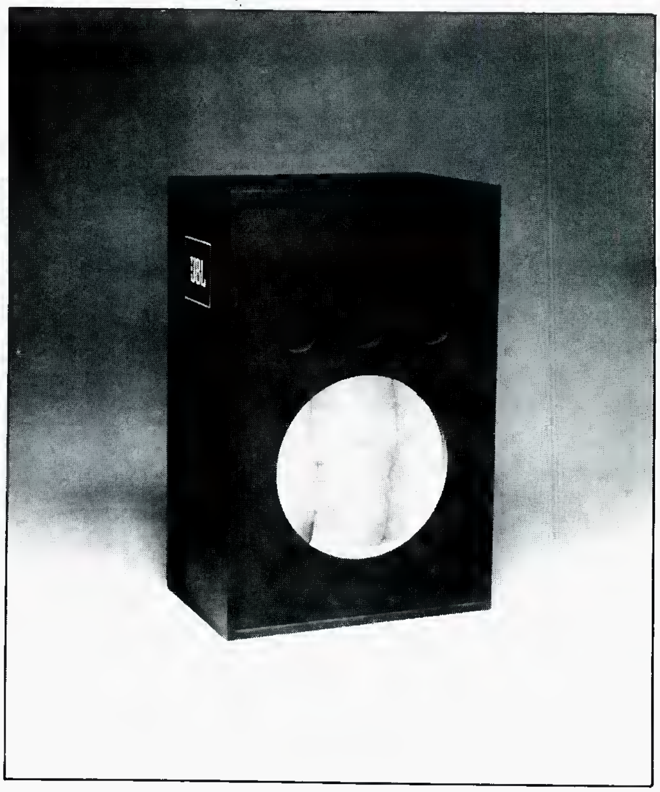
A: Placing two bass drivers in an enclosure designed for one will usually produce less bass and more midrange output, and will upset the operation of the driver-enclosure system because each driver will behave as though it is installed in an enclosure which has only half the internal volume of the original enclosure (with one driver).

[5] Q: What should I do to use two drivers (for more bass)?

A: There are two alternatives. When using two identical drivers, you can build an enclosure with twice the internal volume of the original enclosure that contained one driver, or you can duplicate the original enclosure and stack the two. As the latter alternative suggests, when building the double enclosure, it's necessary to treat the enclosure as if it were two enclosures—you must double the porting used on the single smaller enclosure—although it is not necessary to divide the volume of the double enclosure unless two different driver models (e.g. E130 and E155) are used and their interaction would be undesirable. A usable example of this might be a 227 liter (eight cubic foot) enclosure divided into two chambers so that the E130 occupies fifty-seven liters (two cubic feet) and the E155 occupies 170 liters (six cubic feet). In this case, the ports tuning either chamber to the same desired frequency will be quite different.

[6] Q: What does port or enclosure "tuning" mean?

A: In exactly the same way the resonant note from a bottle can be raised and lowered by adding or pouring out liquid to change the bottle's air volume, enclosure tuning is affected by the ratio of air volumes in the port (the bottleneck) with its attendant flow resistance, and the enclosure interior volume. Tuning of loudspeaker enclosures is a result of manipulating the differences in effective air mass between the enclosure interior and the air in the port. The bottle-like nature of a vented enclosure is known as a "Helmholtz resonator." The ports or ducts in a vented enclosure work only over a narrow band of frequencies near the chosen tuned frequency, producing the same effect noted when blowing across a bottleneck—a single distinct pitch.



JBL Model 4645

[7] Q: Is it always necessary to use a port for good bass?

A: JBL uses vented enclosure designs because they are superior to sealed enclosure designs in several important ways—as long as it is possible to tightly control the loudspeaker driver parameters in manufacturing as JBL does. Vented designs produce lower distortion at the lowest operating frequencies, afford the driver protection against mechanically destructive large cone excursion, and better enable the driver to absorb and utilize its full power rating from an amplifier when operating at low frequencies. It is important to keep in mind that porting and tuning an enclosure provides air loading for the bass driver down to frequencies just below the Helmholtz frequency, but does not provide any loading for the driver at frequencies below that, such as subsonic turntable rumble, record warp, or microphone wind pickup. If you intend to operate a sound system at high power levels, we highly recommend an electronic high-pass filter to eliminate subsonic input to the power amplifier(s). This will substantially increase the available useful power from the amplifier which will then only operate in the audible frequency range. Such a filter is the UREI model 501 Sub Sonic Processor, or the built-in sub-sonic switches of the JBL Electronic Frequency Dividing Network model 5234A.

[8] Q: Where should I locate the port(s) with respect to the woofer?

A: Bass reflex enclosures are usually designed to tune from about 100 Hertz and down. The length of sound waves at these low frequencies is over eleven feet, so port placement is not critical. Ports may be located anywhere on the baffle with no change in bass performance; some designs even locate ports on the back of the enclosure which works well as long as the enclosure is not close to a wall (a couple of port diameters away) and there is an unobstructed air path between the woofer and the port. Overall, it's safest to locate

DESIGN EXAMPLE USING JBL 2235 FIFTEEN-INCH WOOFER:

2235 DATA: $Q_{TS} = .25$ $V_{AS} = 16.2\ ft^3$ $f_S = 20$ hertz

FIND BOX VOLUME FOR FLATTEST RESPONSE - V_B :

$V_B = 15\ (.25)^{2.87}\ (16.2) = 4.55\ ft^3$

Is the box size acceptable ?

If it is, then calculate the system's -3 dB frequency - f_3 by :

$f_3 = .26\ (.25)^{-1.4}\ (20) = 36.2$ hertz

If the f_3 is OK, calculate the box tuning (Helmholtz) frequency by :

$f_B = .42\ (.25)^{-.9}\ (20) = 29.3$ hertz

This box volume will yield the flattest response with no low-frequency passband ripple.

If you want to use a larger or smaller box, calculate the f_3 and f_B using the alternate formula: here are examples using a smaller box of 3 ft³ and a larger box of 10 ft³ with graphs showing the resulting frequency response :

10 ft³ :

$f_3 = (\sqrt{16.2/10})(20) = 25.5$ hertz and $f_B = (16.2/10)^{.32}(20) = 23.3$ hertz

The low frequency passband ripple is then given by:

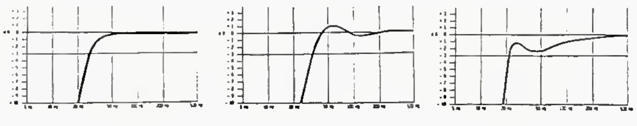
$20 \text{ Log } [2.6\ (.25)(16.2/10)^{.35}] = -2.3$ dB

3 ft³ :

$f_3 = (\sqrt{16.2/3})(20) = 46.5$ hertz and $f_B = (16.2/3)^{.32}(20) = 34.3$ hertz

Ripple in dB $= 20 \text{ Log } [2.6\ (.25)(16.2/3)^{.35}] = 1.4$ dB

FLATTEST DESIGN - 4.55 ft³ UNDERDAMPED - 3 ft³ OVERDAMPED - 10 ft³



the port somewhere on the baffle with the woofer(s) far enough away from side walls to avoid interaction between port and enclosure wall or the fiberglass insulation on the wall.

[9] Q: What should the ducts be made of? Is round better than rectangular?

A: Port ducts may be made of anything rigid, such as paper cardboard with about a 1.5 mm (1/16-in.) or larger wall thickness. They can be any shape, square or rectangular (such that port area remains constant) and made of wood or other suitable material. It is not necessary to use PVC pipe for port tubing, particularly when most carpet stores throw away large amounts of heavy cardboard tubing of between three and four and a half inches inside diameter.

[10] Q: What is the relationship of duct length to port area?

A: When port area is increased, independently of other factors, enclosure tuning is raised. If duct length is increased, independently of other factors, enclosure tuning is lowered. To keep the same tuning (Helmholtz frequency) you will need to increase duct length as you increase port area.

[11] Q: How big should the port be?

A: The bigger, the better. Any port causes some resistance to air movement, and so introduces unavoidable losses in output to the system as a whole. The ratios of port area and length and enclosure volume determine the Helmholtz frequency tuning. Mechanical reactance elements, stiffness, and air mass, control the effective air mass ratios. At very low operating levels, where air in the port does not move very fast, a small short port will behave the same as a large longer port as far as enclosure tuning is concerned. At high power levels, however, the restricted air flow of the smaller port will produce output level losses, some de-tuning and at high enough levels a small port will cause the enclosure to behave like a sealed enclosure with little or no contribution from the port. To minimize resistive losses, the largest practical port should be used. Computer listings of port choices calculated to limit air velocity inside the port duct will list duct sizes which are normally impractical. A 380 mm (fifteen in.) diameter port is not an unreasonable choice for a 380 mm bass driver. However, the necessary length would dictate that such a port might itself have a volume of many cubic feet, sometimes equal to or larger than the original enclosure. A good rule of thumb would be to avoid ports whose circular area is smaller than at least one third the diameter of the driver such as a 127 mm (five in.) diameter port for a 380 mm (fifteen in.) driver. This will usually provide sufficient port area so that the port will not "whistle" when the system is operated at high power levels near the helmholtz frequency—a sure indication of severe system losses and potential power compression and low-frequency output limiting.

[12] Q: Can I use several smaller ports instead of one big one?

A: Yes. However, there is a phenomenon associated with air resistance resulting from air drag on the internal surfaces of port ducts and turbulence at the ends of the ports that requires a duct length correction when several ports are used. For example, when using four 100 mm (four in.) tubes instead of one 200 mm (eight in.) tube (which has the same port area but one quarter the internal surface area), the length needed will be slightly less than that needed for the single 200 mm tube, perhaps five to ten percent less, depending on overall enclosure volume. These effects exhibited by port ducts are exaggerated by proximity of the duct to enclosure interior surfaces or any other type of boundary that may cause air turbulence near the end of the duct. Therefore, it's important to keep duct ends away from the rear of the cabinet or other obstructions by an amount at least equivalent to or larger than the dimension across the port. If you are using a rectangular port that has an enclosure wall as one of its sides, you might have to use some correction.

[13] Q: Is there a simple mathematical way of designing proper enclosures?

A: Yes. A JBL scientist, D.B. Keele Jr., simplified the work of A. Neville Thiele and Dr. Richard Small so that anyone with a pocket calculator and a ruler or straight edge can design the right enclosure volume and choose the right port or duct for a given loudspeaker driver. JBL offers, at no cost, a four page "kit" containing detailed step-by-step instructions, written specifically for non-mathematicians, showing how to use published Thiele-Small driver parameters in enclosure design. Examples are shown with their results graphically represented. An enclosure design flow chart and enclosure venting nomograph are included.

[14] Q: Should the enclosure's baffle be removable?

A: This is a question of mechanical strength and rigidity. All enclosures, particularly those intended for rough portable use, should be constructed with all sides permanently fixed by glue and screws, and sealed air-tight by virtue of well cut and glued joints. It is preferable to mount loudspeakers from the front of the baffle board to eliminate the possibility of reflections from the inside of the loudspeaker mounting hole. It then becomes unnecessary to provide for removing the baffle. Woofer openings are usually large enough to reach through in order to work inside the box, for example, to install other components.

CONVERSION CONSTANTS and USEFUL DATA

LITERS	FEET^3	INCHES^3	METERS^2
1.00 =	.03531 =	61.0 =	.001
28.32 =	1.00 =	1,728 =	.02832
1000.00 =	35.31 =	61,016 =	1.00

MILLIMETERS	INCHES	METERS
1.00 =	.039 =	.001
25.40 =	1.000 =	.0254
1000.00 =	39.370 =	1.000

TO FIND SOUND WAVE LENGTH: divide velocity of sound by frequency (Hz) (SOUND VELOCITY = 344 m/s, 1130 ft/s or 13,560 in/s).

AREA OF CIRCLE = 3.14 x (radius squared). *Note: radius = 1/2 diameter.*

TO FIND THE DIAMETER OF A CIRCLE WITH EQUIVALENT AREA: 2 x square-root of (area divided by 3.14).

Example: area of 9" tube = area of 8" square duct calculated:
(area) 64/3.14=20.37, square root = 4.51 x 2 = 9.03 (diameter).

VOLUME OF CYLINDRICAL DUCT = circular area x length.

VOLUME DISPLACED BY JBL LOUDSPEAKERS: 8" = .05 cu ft, 10" = .1 cu ft, 12" = .15 cu ft, 15" = .2 cu ft, 18" = .3 cu ft.

JBL LOUDSPEAKER MOUNTING HOLE AND BOLT CIRCLE DIMENSIONS:
mounting holes: 8" = 7-1/16" 10" = 9" 12" = 11-1/16" 15" = 13-31/32" 18" = 16-13/16".

Bolt Circles: 8" = 7-5/8" 10" = 9-3/4" 12" = 11-9/16" 15" = 14-9/16" 18" = 17-3/8".

[15] Q: Is there a preferred shape for loudspeaker enclosures?

A: There are a number of shapes that improve performance and some that cause distinct degradation in performance. For single, full-range drivers (e.g. JBL's LE8T) a sphere is the ideal shape for an enclosure because the curved surfaces avoid the diffraction effects of cabinet edges, which bend sound waves in a manner dependent on frequency. For multi-way loudspeaker systems, spheres are usually impractical because of the large size needed and because of the precise orientation required for optimal listening. Conventional enclosures work best mounted flush into a wall where diffraction is controlled by virtue of the wall surface, and for free-standing enclosures, tilting, angled and curving surfaces may be employed to help reduce or control edge diffraction. The overall shape of the enclosure is relatively unimportant except where the shape makes it difficult to build a rigid enclosure. It is best to avoid enclosure dimensions that are multiples of each other, such as 1 X 2 X 4 ratios, and strive to use dimensions that have somewhat unrelated ratios such as 1 X 1.23 X 1.41.

[16] Q: What is the best material to use for building enclosures?

A: For home and permanent installation use, high density particle wood is the most cost-effective material for general enclosure construction. The best wood to use for portable enclosure construction is fourteen to twenty ply per inch Finland birch type. Birch plywood is very expensive, however, and a carefully braced enclosure made of high grade void-free fir plywood can do the job just as well in most cases. The thicker you can make the cabinet walls, the better the results will be because of reduced wall vibration and resonance, but the tradeoff is cost and weight. Enclosure walls should be cut so that edges form an air-tight seal when glued together. Cleats and caulking can also be used if needed to insure a good fit and tight air seal.

[17] Q: Is bracing necessary? How much should be used?

A: Bracing should be added to the enclosure interior to minimize enclosure wall vibration. Enclosure walls simply cannot be stiff enough since wall vibration indicates that energy is being wasted to move enclosure panels rather than moving air. Twenty-five by seventy-six mm (one x three in.) pine bracing fixed on edge with glue and screws to the enclosure walls will help provide the minimum necessary stiffening without affecting the internal volume significantly. If you are building large subwoofer enclosures, bracing with two-by-fours works better, though you should take the bracing volume into account since a 3 m (ten foot) length takes up 12.9 liters (0.36 cubic foot) of enclosure volume.

[18] Q: How should I mount drivers on the baffle?

A: Mount drivers on the front of the baffle whenever possible to avoid the reflections from inside the mounting hole. Heavy drivers should normally be front-mounted using Tee-nuts and machine screws or JBL's MA15 clamps. If Tee-nuts are used, apply a bit of Bostic or Pliobond type rubber glue to the inside of the nut flange to help avoid losing the Tee-nut inside the enclosure when installing the driver. Baffle board construction is much easier if all baffle parts are assembled prior to final box assembly.

[19] Q: Do I need fiberglass inside the enclosure?

A: JBL uses a 25 mm (one in.) padding of one half pound density fiberglass stapled to the enclosure interior on all surfaces except the baffle. You should use 100 mm (four in.) thick dacron or 25 mm (one in.) fiberglass on at least three of the surfaces of parallel interior walls. Keep sound absorbing materials away from the port(s) as the air velocity inside the port can be sufficient to tear off bits of the material and squirt them out of the enclosure. It is not necessary to cover the inside of the baffle, but doing so will rarely degrade system performance. The enclosure exterior may be covered with your choice of any suitable finish or decoration; this will not affect bass performance and in some cases (as with Formica) may help stiffen the enclosure walls.

[20] Q: Does Fiberglass significantly affect enclosure tuning?

A: No, not unless the enclosure is stuffed full of fiberglass, in which case the apparent volume of the enclosure increases by twelve to twenty percent as seen from the point of view of the bass driver. Stuffing the enclosure full with fiberglass is not recommended because it introduces system losses, is expensive and interferes with port operation. The exception to this would be a sealed "air suspension" type system enclosure where more virtual volume is needed and actual volume is not available, and/or where box dimensions which are multiples of each other can't be avoided and the fiberglass stuffing will help absorb the internal sound reflections.

[21] Q: What is needed to mount a midrange on the baffle with the woofer?

A: For cone-type midrange drivers, a sealed sub-chamber should be used to prevent interaction with the enclosure's bass driver. JBL drivers suitable for sealed-chamber midrange use require only ten to forty liters (0.3 to 1.0 cubic foot) of chamber volume to operate at typical midrange frequencies, above 200 Hertz. Subchambers should be constructed solidly and liberally lined with fiberglass. As in the case of enclosure shapes, avoiding multiples of dimensions, subchambers should be built so as to avoid square and cube shapes in favor of non-related numerical ratios.

[22] Q: Is there any special procedure for mounting a horn in an enclosure?

A: Use of a horn/compression driver does not require any subchamber since these devices form their own air-tight seal. JBL horns such as the 2344, 2370, MI-291 and 2380 horn family also seal their own cutout opening in the enclosure when properly mounted on the baffle. Better compression drivers are quite heavy, so a brace should be provided to cradle the driver to prevent driver movement during shipping. In combination with the length of a horn as a lever, driver mass can cause the assembly to tear off the baffle or break the horn if the enclosure is handled roughly or dropped. Driver mass can also tear off the horn throat if cabinets are dropped on their backs.

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USING THE ENCLOSURE DESIGN FLOW CHART

WHAT YOU WILL NEED:

[1] A "scientific" pocket calculator. These can be found at electronic parts stores from about $8. The calculator need have only minimal scientific functions; a "y^x" key, or the ability to raise an input number to the power of an input exponent. The calculator must also have a "LOG" key that

takes the base-10 log of a displayed number, and of course, a square-root key.

[2] A clear plastic straight edge about 6 inches long.

INSTRUCTIONS FOR NOVICES:

The calculations on the flow chart are simple arithmetic. If you use two simple rules, the math will quickly become simple.

[RULE 1] Calculate equations from the inside out.

Look at the equation. Look for the innermost terms. Taking the first equation as our example, $V_B = 15\ (Q_{TS}^{2.87})\ V_{AS}$, the term "$Q_{TS}^{2.87}$" appears inside a pair of parentheses, inside the two other terms in the equation, "15" and "V_{AS}". Using the 2235H as our reference loudspeaker, we first raise its Q_{TS} (.25) to the power of 2.87 to obtain 0.0187, next we multiply by 15 and V_{AS} (16.2), in either order, to obtain 4.55, the box volume in ft³.

Where a term to be operated upon appears under a radical sign (square-root), do the operation first, then take the square-root of the result, then continue with remaining operations. For example, to find the f_3 of a larger or smaller box, the equation used is: $f_3 = (\sqrt{V_{AS}/V_B})(f_S)$. First, divide our V_{AS} (16.2) by the desired box volume, V_B (let's say 3 ft³ for example), to obtain 5.4, then take the square-root of the result to get 2.324, then, finally multiply by f_S (20 hertz) to obtain the answer, 46.5 hertz, which is the −3 dB frequency for a 2235 in a 3 ft³ box.

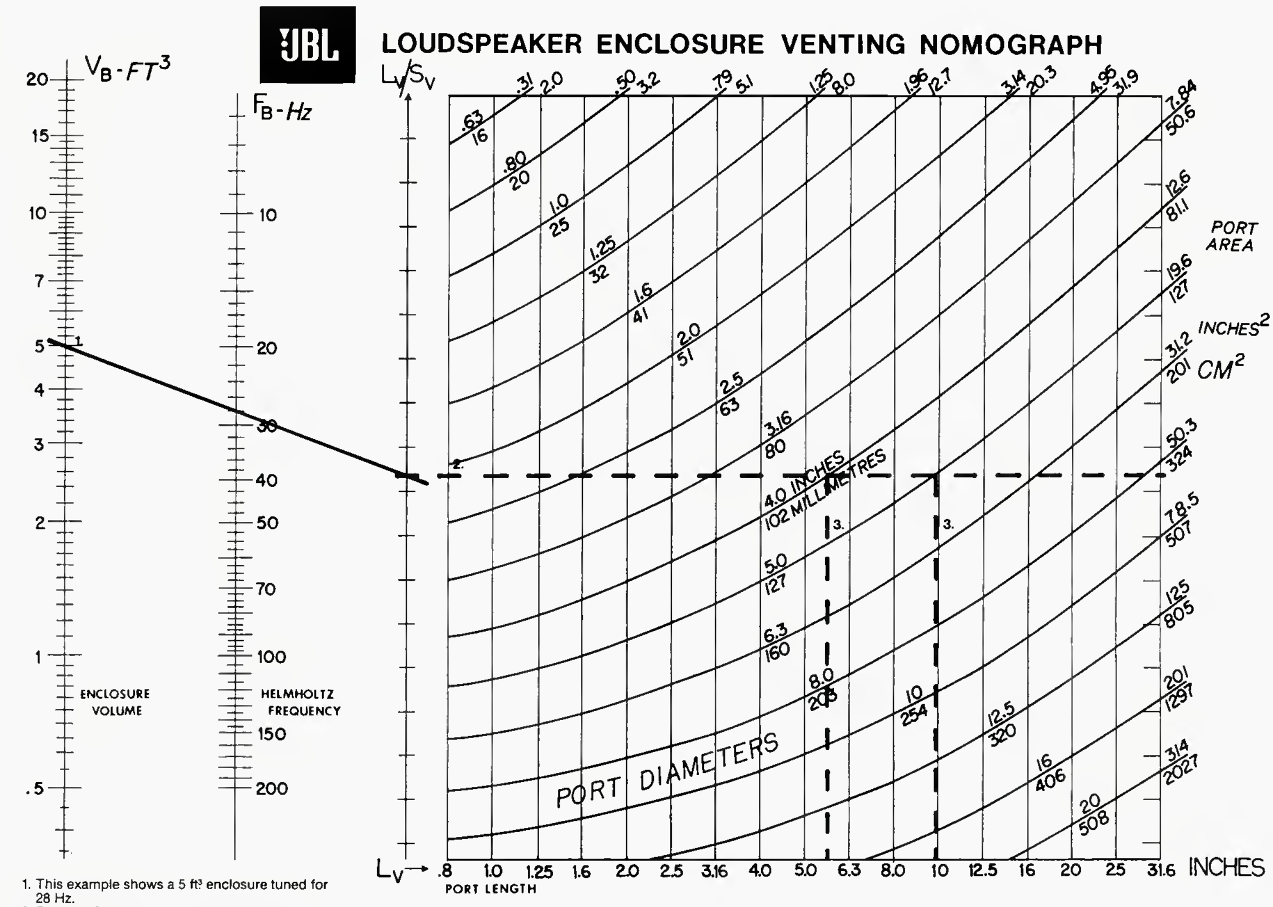
Where two sets of parentheses or brackets appear, the same rule applies. For example, to find the tuning frequency for the oversized or undersized box, use the bottom equation on the page: $f_B = [(V_{AS}/V_B)^{.32}]\ f_S$: first divide V_{AS} (16.2) by V_B; using a 10 ft³ box, 16.2/10, then raise the result (1.62) to the power of .32 to obtain 1.17. Now multiply by f_S (20 Hertz) to obtain 23.3 hertz.

[RULE 2] Ignore the names of the quantities you are calculating.

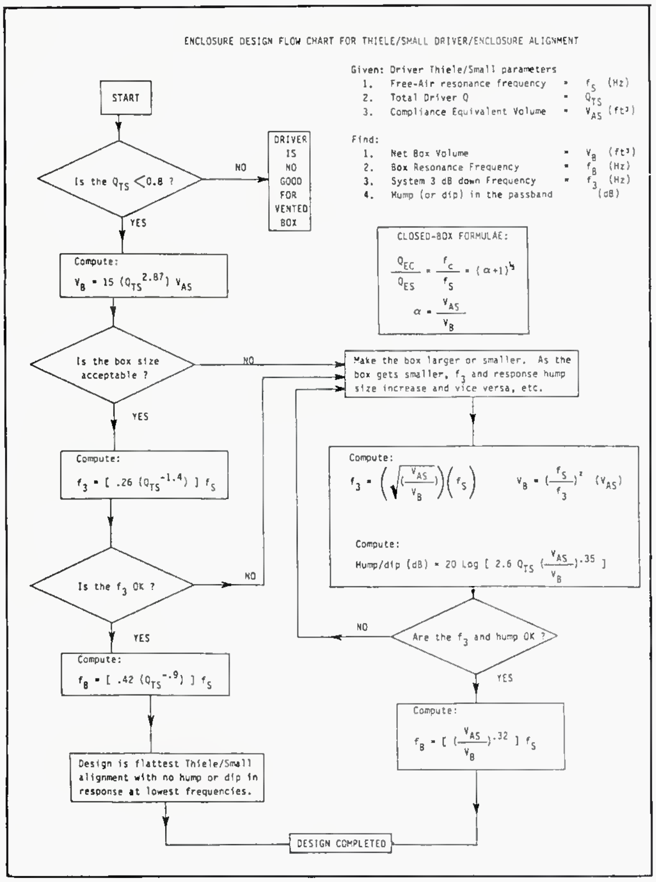
Confusion often arises trying to figure out what to do with the "hertzes" and the "cubic feet," but just remember that the number you end up with at the end of a calculation is the quantity of the term on the other side of the equals sign. As in the case of the last example, the equation is: $f_B = [(V_{AS}/V_B)^{.32}]\ f_S$ so after calculating you will end up with a number that will represent "f_B."

Make notes of your calculated results! Don't go to all the trouble of calculating without noting your results. Make a chart something like this:

	BOX SIZE 3 ft³	BOX SIZE 4 ft³
	f_3 46.5	f_3 40.2
	f_B 34.3	f_B 31.3
hump/dip	dB +1.4	dB +0.5



1. This example shows a 5 ft³ enclosure tuned for 28 Hz.
2. From L_V/S_V construction line, draw horizontally for:
3. Port dimension options. Shown: 4"D x 5½"L and 5"D x 10"L



BOX SIZE 5 ft^3	BOX SIZE 6 ft^3
f_3 36.0	f_3 32.9
f_B 29.1	f_B 27.5
dB −0.2	dB −0.7

After you have determined what box volume you will be using and what f_B you will tune your enclosure to, you will need to know how to port your enclosure to obtain the desired tuning (helmholtz) frequency. Turn to the "Loudspeaker Enclosure Venting Nomograph" and place a straight edge so as to intersect the enclosure volume and helmholtz frequency at the desired points, then make a light pencil mark at the point where the straight edge intersects the "L_V/S_V" construction line. This point is called the "construction point." Next, draw a light pencil line from the construction point, straight through the chart, parallel to the top and bottom. A set of marks with the same spacing as those on the construction line have been added to the right border of the chart to help you to keep the line parallel. Your choice of port diameters and lengths now appears at intersect points along the horizontal line you have last drawn. Use the largest practical port size you can, keeping the end of the port away from inside box walls by at least one full diameter's equivalent distance. When ports require ducts to tune low and maintain large port area, ducts may be round, square or rectangular, and made of any material at least as rigid as thin cardboard carpet tube. Port placement is not critical, since ports only operate at the f_B frequency, which is typically low, producing sound waves which are much longer than loudspeaker-to-port dimensions—at 40 hertz for example, the wavelength is 28 feet.

A NOTE ABOUT SUBJECTIVE RESULTS

TS.BAS (The Basic Program)

```
1 CLS : COLOR 15 : KEY OFF ' IBM functions

2 PRINT : INPUT "DRIVER Fs ";FS

3 PRINT : INPUT "DRIVER Qts ";QTS

4 PRINT : INPUT "DRIVER Vas ";VAS

5 VB = (QTS.87)*15*VAS

6 F3 = (QTS1.4)*.26*FS

7 FB = (QTS.9)*.42*FS

8 PRINT : PRINT USING "BOX VOLUME = ###.## Ft,
F3 = ###.# Hz,  Fb = ###.# Hz";VB,F3,FB

9 PRINT : INPUT "Is the box size acceptable ";A$ :
PRINT

10 IF A$="Y" OR A$="y" THEN 2

11 IF A$="N" OR A$="n" THEN 12

12 INPUT "What box volume would you prefer ";VBX :
PRINT

13 F3X=SQR(VAS/VBX)*FS

14 FBX=((VAS/VBX)32)*FS

15 PRINT USING "For a ###.# ft box, F3 = ###.# Hz,
FB = ###.# Hz";VBX,F3X,FBX

16    L1=LOG(10)    DDB=20*LOG(((VAS/
VBX)35)*QTS*2.6)/L1 : PRINT

17 IF DDB0 THEN B$="+"

18 PRINT USING "The flattest alignment will have
!##.# dB of low frequency response ripple";B$,DDB

19 PRINT "with this box volume" : PRINT : GOTO 9
```

MODEL	fs	Qts	Vas	Eff.	Pmax	Xmax	dia.	Qms	Qes	Re	L
128H	20	.24	9.9	.86%	100	.31	10.2	7.0	.25	5.7	0.
2105H	200	.53	.035	1.2%	25	.06	3.5	3.0	.65	6.1	0.
2108	40	.17	1.2	1.2%	75	.06	6.0	4.5	.18	5.8	0.
2110	60	.31	1.2	2.1%	25	.10	6.5	3.5	.34	6.0	0.
2115	55	.48	1.2	1.0%	30	.22	6.0	4.0	.54	5.5	0.
2118H	85	.35	.5	2.1%	100	.12	6.5	2.4	.40	5.5	0.
2118J	85	.35	.5	2.1%	100	.12	6.5	2.4	.40	10.3	0.
2120	65	.36	1.6	3.0%	75	.06	7.9	4.0	.40	6.0	0.
2121	35	.16	3.9	2.7%	75	.06	7.9	5.5	.17	6.0	0.
2122H	40	.23	2.3	2.4%	100	.12	8.0	1.9	.26	5.8	0.
2123H	75	.27	0.9	3.5%	250	.10	7.9	4.0	.29	4.2	0.
2130	50	.20	4.3	6.9%	100	.06	10.2	4.0	.21	6.3	0.
2135	40	.25	10.5	6.7%	125	.06	13.2	4.0	.27	6.3	0.
2145	30	.51	5.5	.76%	50	.14	9.3	12.0	.53	5.0	0.
2150	55	.64	3.5	2.2%	50	.10	12.2	5.0	.73	5.5	1.
2202H	50	.16	3.1	6.0%	150	.14	10.2	3.5	.17	5.5	1.
2203H	16	.14	14.1	1.1%	100	.20	10.2	6.0	.14	6.3	1.
2204H	45	.35	3.1	1.8%	350	.27	10.3	1.7	.44	6.2	0.
2205H	30	.21	10.5	3.5%	150	.10	13.3	5.0	.22	5.5	1.
2213H	25	.49	8.3	.68%	75	.31	9.8	8.5	.52	4.4	0.
2214H	23	.24	7.9	1.1%	200	.26	10.2	10.5	.25	5.6	1.
2215H	20	.21	26.0	2.6%	100	.16	13.3	5.5	.22	5.7	1.
2220H	37	.17	10.5	8.7%	200	.12	13.3	5.0	.18	5.7	1.
2220J	37	.17	10.5	8.7%	200	.12	13.3	5.0	.18	13.2	2.
2225H	40	.28	6.0	3.5%	200	.20	13.3	2.5	.31	6.3	1.
2225J	40	.28	6.0	3.5%	200	.20	13.3	2.5	.31	12.9	2.
2231H	16	.21	26.0	1.4%	100	.20	13.2	5.5	.22	6.3	1.
2234H	23	.22	16.2	2.1%	150	.33	13.3	2.0	.25	6.0	1.
2235H	20	.25	16.2	1.3%	150	.33	13.3	2.5	.28	6.0	1.
2240G	30	.25	17	5.0%	300	.22	16.0	2.5	.25	2.5	0.
2240H	30	.23	17	5.0%	300	.22	16.0	2.2	.25	6.0	1.
2245H	20	.27	29	2.1%	300	.38	16.0	2.2	.27	5.8	1.
E110	65	.36	1.6	3.0%	75	.10	7.9	4.0	.40	6.0	0.
E120	60	.17	2.8	8.6%	150	.12	10.2	1.8	.19	6.3	0.
E130	40	.19	10.5	8.6%	150	.10	13.3	1.8	.21	6.3	0.
E140	32	.17	10.5	4.9%	200	.14	13.3	5.0	.19	5.5	1.
E145	35	.25	9.7	4.3%	150	.28	13.3	6.0	.26	5.7	1.
E155-4	30	.20	15	4.9%	300	.20	15.0	2.2	.22	2.5	0.
E155-8	30	.20	15	4.9%	300	.20	15.0	2.2	.22	6.0	1.
G125-8	65	.32	2.5	5.5%	200	.10	10.2	5.5	.34	5.2	0.
G135-8	45	.36	8.3	5.5%	200	.10	13.3	5.5	.38	5.2	0.
K110	65	.36	1.6	3.0%	75	.12	7.9	4.0	.40	6.0	0.
K120	50	.20	4.3	6.9%	100	.12	10.2	4.0	.21	6.3	0.
K130	40	.25	10.5	6.7%	125	.12	13.3	4.0	.27	6.3	0.
K140	30	.21	10.5	3.5%	150	.20	13.3	5.0	.22	5.5	1.
K145	35	.29	8.6	3.4%	150	.20	12.5	6.0	.30	8.8	2.
K151	30	.27	12.9	3.4%	150	.20	14.5	6.0	.28	6.0	2.
LE8TH	45	.56	1.2	0.5%	25	.22	6.0	4.0	.65	5.5	0.
LE10H	33	.37	2.7	0.7%	75	.24	7.9	6.9	.39	4.8	0.
LE14H	26	.27	5.2	.89%	150	.33	11.4•	2.3	.30	5.9	1.
LE15A	20	.21	26.0	2.6%	100	.16	13.3	5.5	.22	8.8	2.
MI-10	75	.33	1.3	3.5%	150	.12	8.2	1.8	.41	5.6	0.
MI-12	65	.46	2.7	3.5%	150	.12	10.4	2.2	.58	5.6	0.
MI-15	55	.62	6.0	3.5%	150	.12	13.3	2.8	.79	5.6	0.
MI-15A	40	.42	9.6	3.5%	150	.14	13.3	4.0	.47	5.6	0.

WHAT THIELE-SMALL ALIGNMENT DOES:
Flattest Thiele-Small alignment produces a "system" consisting of box and loudspeaker, which provides the smoothest frequency response, phase response and damping (often called "tightness") that a particular driver is capable of delivering. This system forces the driver to produce minimum distortion and phase shift, and allows the driver to handle maximum power at its lowest operating frequencies.

WHAT ALIGNMENT DOES NOT DO:
Flattest alignment will not allow loudspeakers with small V_{AS} and Q_{TS} and high f_S to produce low bass, because the calculations will place them in small boxes. Drivers with low Q_{TS}, V_{AS} and high f_S are widely used as midrange devices. Midrange drivers usually require relatively small enclosure volumes; porting for such systems does not provide significant improvement in performance over sligtly larger sealed enclosures because the drivers never operate at low enough frequencies to produce significant excursion-related distortion.

Flattest alignment does not always give the lowest possible bass response, as the graphs on the example page show. In fact, you may prefer the sound of non-flat aligned speaker systems, or even non-aligned systems, to that of flat aligned systems, using particular loudspeakers. Some good-sounding examples that illustrate this apparent contradiction are open-back guitar amplifier enclosures, automobile speakers and those vastly oversized hi-fi cabinets built in the Fifties and Sixties to house D130's. The most common use of non-optimum alignments is in musical instrument speaker systems. By Thiele-Small standards, a D130-type speaker is a "midrange" driver, but anyone with a 12 cubic foot D130 enclosure tuned low can tell you that D130's make plenty of bass.

The Thiele-Small flat alignment should be used as a starting point from which to judge other enclosure designs that may produce pleasing or acceptable results with particular loudspeakers. ■

"The Black Art" Gets A Little Closer

Brooke Sheffield Comer

Record mastering steps into the limelight.

(l. to r.): Calbi, Steve Ralbovsky (CBS, Director A&R), Scott Litt (producer/engineer for CBS artist Matthew Sweet). / / C

WHO WOULD think that the advent of compact disks, and a trend toward freelance recording engineers would help to bring record mastering into the limelight? Greg Calbi, of New York's Sterling Sound, explains just how "the black art" has come closer to the forefront of record production than ever before. With CDs creating new potential for quality, and freelance engineers gaining fresh sound perspectives in different studio environments, it makes sense that the role of the mastering engineer is more crucial to the eyes of the industry and the ears of the consumer.

Calbi believes that heightened studio standards over the years have had the biggest influence on the tapes brought in for mastering. "Thirteen years ago when I started mastering, even major studios differed enormously in terms of monitoring systems and board electronics," he reveals. "The tapes I'd get would vary in quality and texture. I'd never know what kind of sound to expect. Because people were more inclined to experiment with monitoring systems ten years ago, some of the tapes I'd get would be drastically lacking in quality. But I think I came into the mastering business at a good time. Sometimes I'd get a tape and literally have to remix it in the mastering room. And once you get a great sounding record from a problem tape, it gives you a feeling of how powerful, and critical the mastering process actually is."

Brooke Sheffield Comer is a freelance writer specializing in audio and video and is the New York editor of On Location Magazine.

Learning to master in the early 70s enabled Calbi to experience greater degrees of freedom to experiment than he's currently allowed. "People were more inclined to admit that their finished product was a problem, and they'd give you more reign to fix it up," he notes. "Not only was there more opportunity for experimentation in the old days, but you were expected to do anything you needed to do to make the tape sound as great as it possibly could."

Because more recording and mixing engineers are working out of a variety of rooms, sound standards are becoming more uniform on the tapes that Calbi receives. "In the last two or three years, as staff engineers have given way to freelancers, there is less dependence on one studio's monitoring system," Calbi observes. "Engineers are using their own gear now. Some even bring their own amps and speakers into a session. I'v found that this is giving the tapes I get a good, standard sound. In the old days when engineers were on staff, they were more isolated and insulated in one room, so their product could be way off. Now they have to be more on the ball. I'm getting tapes now that are much closer in sound quality, though there are trends and tendencies that I can still detect within that quality."

One of the biggest trends shaping the sound of tapes that Calbi receives comes from the SSLs that have taken over many a studio room. "You can use an SSL at Right Track in New York, and finish up your project in Monseurrat," he points out, "and you'll get the same great sound, which makes my work that much easier. The SSL has a sound that mastering engineers get to know. I've learned to work with that sound and understand its problems."

In order to overcome any problems that might arise, Calbi stresses the importance of keeping in close touch with the recording engineer and/or producer throughout the mastering process. "I find it important to connect with the engineer or producer who brings in the tape," he adds. "I like to find out exactly what they're going for in the product. One of my philosophies is that all records shouldn't sound the same, and I'm always willing to go with something that's a little different, but I want the producer or engineer that I'm working with to know what's going on. One of the advantages of being in the business for thirteen years is the perspective I've gained. I can tell someone who wants an inordinate amount of bass on his record that I'll do it but that he'll be getting more bass than ninety-nine of the records I've cut in thirteen years."

With the advent of CDs, Calbi find that mastering techniques are becoming more relevant in the industry, in all product formats. "In the last year, there's been a major conciousness raising in the record companies. Label people are realizing that CDs must sound terrific to sell for fifteen dollars. When a consumer spends that much money, they want a great sound. I know that when I see two CDs put out by different labels, and I know one label has a reputation for spending less time on a product, I won't buy that product. I can't afford a collection of bad-sounding CDs, and I think most consumers can't either."

As CDs come of age, Calbi notes two conflicting trends in the industry that a conscientious mastering engineer can help resolve. "I see producers today realizing that music must sound good in all formats," Calbi observes, "in cassettes, CDs and records. But, by the same token, all the different formats put quality control in too many hands, making it a vulnerable situation for the producer. That's why you'll see a lot more clauses in contracts in the future, stipulating producer's control over all formats of the final product. If I were to produce, I'd certainly insist on that. A producer's next job depends on the sound quality of the consumer product. And if someone at the label didn't do his job properly, after six months of work, I could end up with a product that isn't as great as it could be, and I might not get that next job."

The solution to better sound, Calbi explains, can often come from the middle man—the mastering engineer. "Engineers and producers must work together and assert themselves so that the labels will guarantee proper work on the product. It's not so much a matter of money as it is of time. Release schedules for CDs are usually behind deadline. You'll notice CDs come out as long as six to seven weeks after the release of the record, which means that labels are having a hard time keeping up with production schedules. The remedy lies in coordination rather than cash. Part of the coordination involves an inter-relationship between producer, A&R man, and engineer. I assert myself as much as I can with the label and the producer, and I find it's amazing how accomodating people can be when a situation is explained."

Mastering CDs has made Calbi more attuned to what makes a good sound disk. "In disk mastering, engineers have to correct a lot of monitor errors," he reveals. "And today, a lot of compilation records are coming out with multiple producers and extra tracks from different sources, so I'm finding a lot of diversity. This kind of work lends to the essential goal of mastering; bringing sounds out and making levels compatible. CDs really accentuate that goal, especially in the re-issuing of old masters. I'm going to do a 12-record series of Stones re-issues on Columbia this summer, and we're going to research every possible source. We'll get as many tapes as we can find to establish maximum compatibility. That way we'll get the best sound we can possibly find. Projects like the Stones are very time consuming and expensive, but to compete in today's market, and create a CD with fifteen dollars, you have to have something fantastic."

Calbi cites Prince's *1999* as an example of dynamic range. "That CD is so clean sounding that the record sounds a little brittle in comparison. There's a case of a more enjoyable CD, which is the point of them. That's why they cost more. The CD is here to stay. There's no bucking it, and no reason for engineers to fear it. It's just re-awakened people to what a sonically good record should be. And there's a new invention now, a laser oriented playback that plays regular analog records back. It's in the $2500 range, which isn't affordable for the average consumer, but I'd love to see it take off because there are some advantages to the analog disk if you can get over the vinyl and noise problems. That can be exciting."

Calbi uses a Neve Transformalist Disk Mastering Console "that was custom designed to accomodate to my own special layout of the room," he explains. "The console was a recent addition, and it's amazing what a difference it makes. A lot of engineers know the rich, round sound of Neve. It's always seemed to put a top end on a sound that you can hear on a home system."

"My old console was a 1966 Telefunken, "Calbi continues, "and it was interesting to compare to two. When (producer) Jim Steinman and I finished the final EQ on a Bonnie Tyler album, and had completed the final mastering, the console was replaced. We wondered what the EQ would sound like on the new console, and found that though it had a completely different texture, the sound came very close. The Neve gives me more flexibility though, and a more modern sound, though I can't help missing the three dimensional top end on the old board. That's something I'll miss."

Direct Metal Mastering is another exciting new facet of the business for Calbi. "Sterling got their DMM system in last November," he recalls, " and (chief engineer) Ted Jensen has been working hard to tune up the system. Though the system is sounding good, it's not a dramatic improvement. It has, however, been effective in making label people aware of what good mastering is, and what it can do. By insisting on a certain quality of vinyl, TELDEC (DMM's organization), insures pressings of high quality. If the vinyl is bad, you'll have a gritty and noisy sounding record no matter how well the mastering engineer does his job." Calbi finds himself putting "a touch more low end on records now because DMM seems to give the top end a silky, smooth sound. But the truly amazing thing about DMM is the subtle opening of the mid range, and the very clean middle that consumers will hear. It's a step in the right direction, this emphasis on quality, and everyone is enjoying the results.

With so many formats coming out now, mastering houses, Sterling included, have been busier than ever. "We can't keep up with all the record and CD mastering that we've been doing," he explains. "Promotional seven- and twelve-inch mixes have always been an important part of our business. We've also doing considerable work on four-track for TV music videos," Cabli notes. "A lot of people play their MTV through their speakers, and every little edge in the competitive market is going to help sales." As new strides in sound quality are continually forged, mastering engineers will enjoy a more visible, and audible place in the production arena. ■

db Buyer's Guide

Reverbs & Delays

Equalizers

Crossovers

DELAY AND REVERB

ADA SIGNAL PROCESSORS
The STD-1 Stereo Tapped Delay generates true stereo outputs from a mono source through six simultaneous user-selectable delay taps. It produces multi-voice chorusing, true stereo flanging, and psycho-acoustic studio effects. Dimensions are 1.75 x 19 x 10.5; weight is 6.5 lbs.
Price: $799.95.

The Digitizer 4 Programmable Digital Delay allows complete programmability of all effect settings including sweeps, mix and regeneration. It has 16 user programs and 16 factory preset programs that are randomly accessible. Dimensions are 1.75 x 19 x 10.5; weight is 6.75 lbs.
Price: $699.95.

The 2FX Digital Multi-Effects produces two independant effects simultaneously. Separate groups of controls allow the user to preset digital flange, digital chorus, and digital delay. Features over 1 second of delay with a bandwidth of 17 kHz. Dimensions are 1.75 x 19 x 10.5; weight is 6.75 lbs.
Price: $599.95.

AKG ACOUSTICS
The BX25E stereo reverb uses AKG's patented Torsional Transmission Line principal to create full, smooth and natural sounding reverberation. Features include continuously variable reverb time, adjustable balanced inputs and outputs, remote control and equalization. Dimensions are 21 x 20 x 18; weight is 66 lbs.
Price: $5,500.00.

The BX25ED provides all the features of the BX25ED, above, with the addition of a digital delay. This provides pre-delay and 2 echoes per channel to enhance the sound field. It has separate outputs for delay. Dimensions are 21 x 20 x 18; weight is 74 lbs.
Price: $8,400.00.

The ADR 68K is a remote controlled digital reverb, with full sampling and special effect capabilities. It has full MIDI interface with dynamic parameter control, stereo splits, and 8 seconds of sampling at 15 kHz bandwidth which can be broken up into 4 segments. Front panel RAM cartridge allows user memory banks to be easily transported. Dimensions are 2 x 19 x 13 (remote is 11 x 8.5 x 2.4); weight is approx. 18 lbs.
Price: $4,995.00.

The TDU-8000 is a modular digital delay system configurable with up to 2 inputs and 8 outputs. Delay times are programmable from 0.1 ms to 1300 ms (650 ms in 2-channel mode) for each output. Ten programs can be stored on EE-Prom non-volitle memory. It has 20 kHz bandwidth with 100 dB dynamic range. Dimensions are 3.5 x 19 x 16.2; weight is 15 lbs.
Price: Depends on configuration.

The MSP-126 is a multi-tap stereo processor that produces stereo from mono sources and left to right image positioning through time delay. Also contains digital reverb early reflection program, 20 kHz digital delay line (376 ms maximum), backwards tape effect, stereo repeats, etc. Dimensions are 3.5 x 19 x 11; weight is 13 lbs.
Price: $1,195.00.

ALESIS

The MIDIVERB is a controllable digital reverb featuring 63 preset programs, and 9 gated and 4 reverse reverb times.
Price: $399.00.

The XT:c is a 16 kHz bandwidth delay with reverse and gated reverb programs, and decay times ranging from 0.1 to 12 seconds.
Price: $749.00.

APPLIED RESEARCH & TECHNOLOGY (ART)

The ART 1500 ms is a digital delay with a delay range of 0.15 ms to 1500 ms in 4 ranges. It has 20 kHz bandwidth at all delay settings, 10:1 sweep ratio, front and rear infinite repeat, 90 dB dynamic range, and 0.2% maximum distortion. Dimensions are 1.75 x 19 x 6.25; weight is 5 lbs.
Price: $500.00.

The ART PD3 is a delay system with 3 outputs, with each output individually adjustable in precise 1 millisecond increments to a maximum of 255 milliseconds. Dimensions are 1.75 x 19 x 10; weight is 9 lbs.
Price: $749.00.

The DR1 is a programmable digital reverb with performance MIDI control. It has 5 each plate, room, and hall algorithms plus gated, reverse, DDL, flange/chorus, and special effects. Dimensions are 1.75 x 19 x 9; weight is 10 lbs.
Price: 1,295.00.

The DR2a is a programmable 3-preset digital reverb system that includes 3 plate, 2 room, 2 hall, gated and reverse algorithms. There are 7 user adjustable parameters including pre-delay, high frequency damping, position, diffusion, and decay time. Dimensions are 1.75 x 19 x 9; weight is 9 lbs.
Price: $595.00.

ARIA MUSIC

The GR 535 is a gated stereo spring reverb with dimensions of 3 x 19 x 8.
Price: $549.00.

The AR 525 is a stereo spring reverb with dimensions of 3 x 19 x 8.
Price: $349.00.

The DEX 100 is a digital delay with dimensions of 3 x 19 x 8.
Price: $299.00.

AUDIO + DESIGN

The Scamp S24 Time Shape Module features a main and auxiliary mix input, positive and negative flanging/phasing, and automatic double tracking. The delay can be manually set or controlled by an envelope follower. A modulation control varies the delay swing about the predetermined center, while a frequency control determines the rate of swing over the delay range (for Doppler and siren effects).
Price: $580.00.

BIAMP SYSTEMS

The MR-140 is a mono spring reverb with dual-stage input limiter, equalizer blend controls, and reverb pre-emphasis. It has balanced and unbalanced XLR and 1/4-in. phone connections, footswitch jack, ground strap, and THD of 0.01%. Dimensions are 1.75 x 19 x 9; weight is 7.25 lbs.
Price: $349.00.

DELTALAB

The DeltaLab ADM 465 Triple Tap Digital Time Delay is designed for the sound contracting market. Features include 20 Hz to 20 kHz bandwidth, 90 dB dynamic range, XLR inputs and outputs, and 155 ms of delay at each tap adjustable in 5 ms increments for a possible 465 ms of maximum delay. The 465 employs Adaptive Delta Modulation (ADM), a proprietary technolgy which DeltaLab pioneered. Dimensions are 1 3/4 x 19 x 7; weight is approx. 10 lbs.
Price: $899.00.

DIGITECH

The RDS-6400 is a digital reverberation system offering 64 different combinations of parameters of reverberation: room/plate, size, decay time, and high frequency damping. The 16 bit A/D/A conversion yields quiet operation and proprietary programs offer natural sounding reverberation.
Price: $599.95.

The RDS-3.6 is a digital delay offering up to 3.6 seconds of delay at half bandwidth (8 kHz) with a 10:1 flange ratio. It features chorusing, vibrato, doubling, echo, multiple echo, flanging, feedback with flanging, comb filtering and infinite repeat. Dimensions are 1.75 x 19 x 8; weight is 7 lbs.
Price: $259.95.

The RDS 1900 is a digital delay offering approximately 2 seconds of full bandwidth (15 kHz) delay with a 10:1 flange ratio. It has the same features as the RDS-3.6, above. Dimensions are 1.75 x 19 x 8; weight is 7 lbs.
Price: $279.95.

The RDS 2001 digital delay offers up to 2 seconds of full bandwidth delay with sampling and multiple footswitch controllable functions. It has the same features, dimensions and weight as the RDS 1900, above.
Price: $299.95.

The RDS 3600 digital delay offers up to 7.2 seconds of delay in 3 ranges (each higher range reduces bandwidth) with a 10:1 flange ratio. It has the same features, dimensions and weight as the RDS 2001, above.
Price: $399.95.

DOD

The R-845 is a spring reverberation unit with range, input and output controls, a 4-band equalizer, and mix control. It has prelimiting and mild precompression to reduce boings and twangs from drum tracks and other sharp percussive signals. Dimensions are 1.75 x 19 x 6; weight is 5.5 lbs.
Price: $249.95.

The R-848 reverb uses 6 pairs of springs, each with a different time constant. It features variable pre-delay to boost room size when necessary, 2-bands of quasi-parametric equalization for fine tuning, and a drive control to optimize signal to noise ratio. Dimensions are 1.75 x 19 x 9.75; weight is 7 lbs.
Price: $399.95.

ELECTRO-VOICE

The EVT 4500 is a 5-spring reverb unit with variable reverb time, 3-band equalizer, input level control, LED level display, floating threshold compressor/limiter, and reverb/direct mix control. Dimensions are 1.75 x 19 x 12; weight is 9 lbs.
Price: $498.00.

EVENTIDE

The SP2016 effects processor/reverb is a full stereo reverb and effects unit that has a variety of reverb programs including, stereo room, plate, hi-density plate, reverse, nonlinear, and multi-tap delay. Optional channel vocoder, automatic panner, and stereo synthesis effects are also available. Dimensions are 3.5 x 19 x 12; weight is 13 lbs.
Price: $6,895.00.

The H969 ProPitch Harmonizer features pitch change, delay, and pitch change presets, front panel preamplified input, flanging and doppler modes. Dimensions are 3.5 x 19 x 12; weight is 13 lbs.
Price: $4,500.00.

The H949 Harmonizer features pitch change (1 octave up and 2 octaves down), delay, flanging, reverse audio, repeat, time reversal, random delay and time compression capability. (Keyboard also available.) Dimensions are 3.5 x 19 x 11.75; weight is 13 lbs.
Price: $3,500.00.

The H910 Harmonizer is a pitch change/effects unit that can be used for doubling vocals, delay and many other special effects, including many types of reverb and echo. Dimensions are 3.5 x 19 x 9; weight is 11 lbs. (Keyboard also available.)
Price: $1,500.00.

The JJ193 digital delay has 4 outputs and 1 input and is available in 510 ms, 1.022 second and 2.046 second versions, set by front panel DIP switches. Dimensions are 1.75 x 19 x 8.5; weight is 5 lbs.
Prices: 510 ms: $1,195.00; 1.022 sec: $1,295.00; 2.046 sec: $1,495.00.

The CD254 digital delay has 2 outputs and 1 input, and 254 ms of delay, set by internal DIP switches. Dimensions are 1.75 x 19 x 8.5; weight is 4 lbs.
Price: $895.00.

FOSTEX

The 3180 is a 3-spring stereo reverb with 24 ms pre-delay, decay time of up to 3 seconds, RCA and 1/4-in. phone jacks, and unbalanced connectors. Dimensions are 3.5 x 17 x 8.25.
Price: $400.00.

FURMAN SOUND

The RV-3 is a digital reverb system with 512 distinct program settings. Included are 2 types of plates, 2 room sizes, and 2 hall sizes. Other controls include pre-delay (50 ms), front/rear position, and 4 decay time settings per room. Dimensions are 1.75 x 19 x 8; weight is 9 lbs.
Price: $599.00.

The RV-1 is a single channel spring reverb system that includes 2-band equalization section with mid-frequency sweep control. It uses 3-spring Accutronics Reverb System for smooth, natural sound. It has ground lift switch, and external transformer. Dimensions are 1.75 x 19 x 8; weight is 7 lbs.
Price: $321.00.

The RV-2 is the same as the RV-1, above, but it is a 2-channel version. Weight is 9 lbs.
Price: $505.00.

GOTHAM/EMT

The EMT 445 is a digital delay system with 1 or 2 inputs, 2, 4, or 6 outputs, 20 kHz bandwidth at all delay times, and up to 10.9 seconds of delay in stereo. It is remote controllable and rack mountable. Dimensions are 7 x 19 x 18; weight is 30 lbs.
Price: $6,390.00.

The EMT 252 is a digital reverberation system that is rack mountable, with remote control and memory of presets. It includes EMT 250 reverberation program, 16-bit, 34 kHz sampling and processing, and individual controls for reflections. Dimensions are 7 x 19 x 18; weight is 35 lbs.
Price: $18,000.00.

The EMT 240 Gold Foil reverberation system uses a gold foil medium with stereo in and out, and adjustable reverb time from 0.7 to 5 seconds. Dimensions are 25 x 12 x 25; weight is 148 lbs.
Price: $9,660.00.

HOSHINO

The SDR 100 is a stereo digital reverb with 30 presets, 70 programmable preset locations, and it is 16-channel MIDI controllable. It has 2 independent processors, 4-band programmable equalizer, and 16-bit linear/parallel processing.
Price: $895.00.

The DD 700 is a digital delay with 12 kHz bandwidth and over 100 ms delay. It has modulation sweep ratio, advanced noise reduction circuit and full stereo processing.
Price: $260.00.

The DD 1000 is a dual digital delay with two independent stereo delays in one package. Channel A provides over 250 ms of delay with 8:1 modulation sweep ratio, and channel B provides over 100 ms of delay. Both have advanced noise reduction circuit.
Price: $399.00.

The DMD 2000 is a programmable digital delay with 8 programmable presets, over 2000 ms of delay and 16 kHz bandwidth. It has full modulation control and optional remote footswitch.
Price: $599.00.

JBL/UREI

Model 7922 Audio Delay. 0.2 ms to 327 ms audio delay in 10 microsecond steps. Two independently delayed outputs. 16 bit linear conversion, 90 dB dynamic range. Linear phase anti-aliasing filters. Digital oversampling filter. Simple headroom control and indicator. Active balanced inputs, transformer outputs. Key lock-out security. Delta A-B display/control for precise arrary alignment. Low noise A-D converter. Single rack space package.
Pricing and dimensions were not available at press time.

KLARK-TEKNIK

The DN780 reverberator/processor is a highly advanced digital system combining room simulation, plates, and effects in one package. Reverberation modes include normal, non-linear, and gated decay, while the effects processing includes ADT, multi- tap echo, and infinite room. Remote control is included. Dimensions are 3.5 x 19 x 12.25; weight is 16.5 lbs.
Price: $3,900.00.

The DN716 is a digital delay system utilizing 16-bit linear A/D conversion providing 20 Hz-20 kHz bandwidth with 90 dB of dynamic range. The 3 outputs are independently adjustable from 0 tp 1.3 seconds in 20 microsecond steps. The input level, 3 output levels, LED readout and headroom indicator are on the front panel. Dimensions are 1.75 x 19 x 11.75; weight is 6 lbs.
Price: $1,625.00.

LEXICON

The PCM60 is a digital reverberator with dimensions of 1.75 x 19 x 11, and a weight of 9.2 lbs.
Price: $1,040.00.

The PCM70 is a 6-voice digital effects processor/reverberator with dimensions of 1.75 x 19 x 13.5, and a weight of 10.7 lbs.
Price: $2,295.00.

The Model 200 is a digital reverberator with dimensions of 5.25 x 19 x 15, and a weight of 18 lbs.
Price: $4,800.00.

The PCM41 is a digital delay processor with a delay time of 800 ms. Dimensions are 1.75 x 19 x 11; weight is 5.5 lbs.
Price: $715.00.

The PCM42 is a digital delay processor with a delay time of 2400 ms. Dimensions are 1.75 x 19 x 11; weight is 5.5 lbs. Also available with MEO and a 4800 ms delay time.
Price: $1,000.00; $1,235.00 w/MEO.

The Model 95 Prime Time II is a digital delay processor with a delay time of 1.92 sec. Also available with MEO-1 (3.84 sec. delay time) and MEO-2 (7.68 sec. delay time). Dimensions are 3.5 x 19 x 13.5; weight is 10.5 lbs.
Price: $1,980.00; $2,200.00 w/MEO-1; $2,500.00 w/MEO-2.

The Model 97 Super Prime Time is a digital delay processor with programmable 0.96 sec. digital delay time. Also available with MEO with 1.92 sec. digital delay time. Dimensions are 5.25 x 19 x 13.5; weight is 17 lbs.
Price: $3,170.00; $3,390.00 w/MEO.

LT SOUND

The ECC Echo Control Center is a digital delay system also having Microplate reverb capability. The delay and reverb can be used together or independently. Delay times are from 1 millisecond to 1 second. Delay time on reverb is variable from 0.6 sec. to 2.4 sec. Effects include doubling, chorus, flanging, plate reverb with delay, acoustic chamber and tremolo. Dimensions are 1.75 x 19 x 7.5.
Price: $995.00.

The RCC Reverb Control Center is a complete MicroPlate reverb system for use with or without a mixing board. It has 2 mic inputs, inputs for 2 additional stereo sources, and an output for a tape recorder. It has 3-band equalizer. Dimensions are 1.75 x 19 x 7.5; weight is 7 lbs.
Price: $595.00.

The RV-2 Stereo Reverb Unit features the MicroPlate reverb system and has over 18 kHz of frequency response. It has 4 simultaneous inputs per channel for 3 different sounds, 7 segment LED level indicator on each channel, and decay time control of 0.6 to 2.4 seconds. Dimensions are 1.75 x 19 x 1.75; weight is 8 lbs.
Price: $895.00.

ORBAN

The 111B is a dual channel spring reverb with six springs/channel for smoothness and natural sound. It has shelving bass and quasi-parametric midrange equalizer, balanced and unbalanced inputs, and transformer balanced main output. Dimensions are 3.5 x 19 x 12; weight is 10 lbs.
Price: $899.00.

PEAVEY ELECTRONICS

The DEP 800 Digital Effects Processor provides continuously variable delay settings from 1.6 milliseconds to 800 milliseconds. It has high and low level inputs and stereo line level outputs, modulation depth and speed controls (0.1 Hz-10 Hz; 4:1 maximum range). Mounts in one 1.75 x 19-in. rack space.
Price: $349.50.

The DEP 1310 Digital Effects Processor provides continuously variable delay settings from 0.5 to 1310 milliseconds. It has high and low level inputs and stereo line level outputs, modulation and depth speed controls (0.1 Hz-10 Hz; 20:1 maximum range). Mounts in two 1.75 x 19-in. rack spaces.
Price: $649.50.

The PEP 4530 Programmable Effects Processor provides continuously variable delay settings from 0.1 milliseconds to 4095 milliseconds. All operating parameters are programmable. It has a storage capacity of 10 factory input programs and 520 user input programs, digital readout for delay time setting and MIDI channel select. Mounts in one 1.75 x 19-in. rack space.
Price: $699.50.

RANE

The Model AD 13 Digital Delay is a 1 input, 3 output delay system for permanent sound system alignment. It features 16-bit microprocessor, 16 MHz clock, 50 kHz sampling rate, up to 320 ms delay, channel select, 4-segment input level meter, separate level controls for all inputs and outputs and 3-digit time display.
Price: $999.00.

ROLAND

The SRV-2000 is a MIDI compatible digital reverb that provides 15 preset room settings in addition to 64 user programmable settings which can be recalled by a MIDI program change command. It has 16-bit A/D/A, a frequency response of 30 Hz-10 kHz, a S/N ratio of 80 dB, a dynamic range of 90 dB, and reverb time of 0.1 to 99 seconds. Dimensions are 2 x 19 x 14.25; weight is 11.5 lbs.
Price: $1,295.00.

The SDE-3000 is a digital delay that provides delay times of up to 4.5 seconds. It has 16-bit A/D/A, a dynamic range of 100 dB, THD of less than 0.03%, a frequency response of 10 Hz-17 kHz with the delay range set from 0 to 1500 ms, S/N ratio of 88 dB, and 8 programmable memories. Dimensions are 1.9 x 19 x 11.8; weight is 11 lbs.
Price: $1,195.00.

The SDE-2500 is a MIDI compatible digital delay offering 64 programmable memories which can be recalled by a MIDI program change command. Maximum delay time is 750 ms, and it has 15-bit A/D/A, 96 dB dynamic range, S/N ratio of 84 dB, and frequency response of 10 Hz-17 kHz with a delay time set from 0 to 375 ms. Dimensions are 1.9 x 19 x 11.8; weight is 10 lbs.
Price: $750.00.

The SDE-1000 digital delay uses a 12-bit A/D/A conversion system with logarithm compression to provide a dynamic range of over 90 dB with THD less than 0.08%. Frequency response is 10 Hz-17 kHz with delay times of 0 to 375 ms. Maximum delay time is 1,125 ms, and 4 battery backed-up programmable memories are provided. Dimensions are 1.9 x 19 x 11.8; weight is 11 lbs.
Price: $595.00.

The RDD-10 digital delay provides delay times of 0.75 to 400 ms. It has a 12-bit A/D/A conversion system with logarithm compression, a frequency response of 20 Hz-15 kHz and mix and delay outputs. Dimensions are 1.8 x 8.6 x 6.7; weight is 2 lbs.
Price: $290.00.

The RSD-10 is a digital sampler/delay with the same conversion system as the RDD-10. Maximum sample or delay time is 2,000 ms with a bandwidth of 20 Hz-7 kHz. Samples can be auto-recorded or manually recorded using a trigger input. Dimensions are 1.8 x 8.6 x 6.7; weight is 2 lbs.
Price: $325.00.

The DSD-2 is a compact effects pedal which can be used either as a digital sampling unit with a sample time of 200 to 800 ms, or as a digital delay with delay times of 50 to 800 ms. A 12-bit A/D/A converter ensures frequency response of 40 Hz-7 kHz. Dimensions are 2.2 x 2.8 x 5; weight is 1 lb.
Price: $325.00.

The DD-2 digital delay offers adjustable delay times of 12.5 to 800 ms in a compact pedal. Same converter and frequency response as the DSD-2. It has 2 output jacks. Dimensions are 2.2 x 2.8 x 5; weight is 1 lbs.
Price: $275.00.

The DM-3 is a compact pedal delay with delay times from 20 to 300 ms. It has 2 output jacks for stereo effects. Dimensions are 2.2 x 2.8 x 5; weight is 1 lb.
Price: $210.00.

ROSS SYSTEMS

The DDL-1000 is a 14-bit resolution, 1024 ms digital delay with modulation depth and rate control, in and out of phase effect, variable output, and hold function.
Price: $349.95.

The DDL-999 is a 12-bit resolution, 1024 ms digital delay with modulation control, in and out of phase outputs, and mix control.
Price: $249.95.

The MM-99 is a digital sound processor with pitch transposer, reverb, delay and surround sound, delay range of 0 to 250 ms, reverb level control, and separate mic and line inputs.
Price: $795.00.

WASHBURN

The WD-700 digital delay offers 640 ms of delay, flanging, phasing, chorus, reverb, doubling, and slapback effects. It utilizes full 16 kHz bandwidth. Dimensions are 1 x 19 x 8; weight is 1 lb.
Price: $349.00.

YAMAHA

The REV1 is a digital reverberator for use in recording, broadcast, film and sound reinforcement. The frequency response is 30 Hz-18 kHz, and dynamic range is 90 dB. Memory 1-30 in ROM and 31-99 in RAM. Remote control included. Dimensions are 5.25 x 19 x 15; weight is 21.25 lbs.
Price: $13,500.00.

The REV7 is a versatile high quality digital reverb/signal processor providing up to 15 early reflections, 99.9 ms of initial delay and up to 9.9 seconds of subsequent reverb. It provides mixing of direct and reverb signals. Dimensions are 3.5 x 19 x 13.5; weight is 11.5 lbs.
Price: $1,325.00.

The SPX90 is a digital multi-effects processor providing echo, reverb, delay, pitch change, gating, reverse gating, parametric eq, sampling, modulation, vibrato, auto-pan, compression, flanging, chorus, phasing and more. Dimensions are 1.75 x 19 x 11.25; weight is 7 lbs.
Price: $745.00.

The YDD2600 is a digital delay system for sound reinforcement, film and video production, recording, and disc mastering. Frequency response is 20 Hz-20 kHz, dynamic range is 90 dB, sampling frequency is 48 kHz. Remote control unit supplied. Dimensions are 5.25 x 19 x 15; weight is 22.25 lbs.
Price: $8,350.00.

CROSSOVERS

ALTEC LANSING
The Model 1631A is a single-channel, 2-way active crossover with user selectable crossover frequency from 100 Hz to 8 kHz. Filters are maximally flat 18 dB/octave Butterworth type. It also has low-pass delay control, high-pass gain, and eq controls. Dimensions are 1.75 x 19 x 5; weight is 4.75. lbs. Price: $552.00.

AUDIO + DESIGN
The SCAMP S27 is a 4-way crossover and summing amp. It has 4 split bands (220 Hz, 1.6 kHz, and 4.5 kHz), and proprietary phase compensated circuitry.
Price: $350.00.

AUDIO LOGIC
The X-324 is a stereo 3-way, stereo 2-way with mono sub-woofer, or a mono 4-way crossover. It has 18 dB/ocatve Butterworth filters with a switchable 40 Hz high-pass filter. All connections for stereo to mono mode switching are internal and require no patching or rewiring. Dimensions are 1.75 x 19 x 8; weight is 6.3 lbs.
Price: $329.95.

BIAMP SYSTEMS
The MX-2 is a mono 2-way crossover with signal/peak indicators, high-pass filter and high frequency phase switch, floating and balanced and unbalanced XLR and 1/4-in. phone connectors, ground strap, THD of 0.01%, and noise level of -85 dBm. Dimensions are 1.75 x 19 x 6; weight is 4.5 lbs.
Price: $299.00.

The SX-23 is a stereo 2-way, mono 3-way crossover with the same features and specifications as the MX-2. Dimensions are 1.75 x 19 x 6; weight is 5 lbs.
Price: $499.00.

The SX-35 is a stereo 2 or 3-way, mono 4 or 5-way crossover with the same features and specifications as the MX-2. Dimensions are 1.75 x 19 x 6; weight is 6 lbs.
Price: $599.00.

BROOKE SIREN SYSTEMS
The FDS360 integrated frequency divider/limiter is a stereo 2- way or mono 3-way or 4-way crossover incorporating mid-filter speaker protection limiters in each section, individual section mute, and individual section gain control. There is also a 0 to 360 degree phase adjustment control at each crossover point. Dimensions are 1.75 x 19 x 9; weight is 10 lbs.
Price: $1,025.00.

CARVIN
The XC-1000 is a stereo active crossover that utilizes a switch that makes it internally tri-ampable. It features extremely low noise, 18 dB/octave fully parametric filters, 112 dB S/N ratio, and a THD of 0.01%. Dimensions are 3.5 x 19 x 6; weight is 11 lbs.
Price: $279.00.

DOD
The R-835 is a stereo 2-way or mono 3-way crossover with 18 dB/octave state variable Butterworth filters. This configuration assures symetry about the crossover point generating both high pass and low-pass outputs simultaneously. Dimensions are 1.75 x 19 x 6.5; weight 4.5 lbs.
Price: $259.95.

ELECTRO-VOICE
The XEQ-3 is a mono 3-way electronic crossover/equalizer incorporating fourth-order Linkwitz-Riley frequency dividing networks for 24 dB/octave slopes. It has variable time delay on all outputs, plug-in equalizer modules using low eq for infrasonic filtering, and level display. Dimensions are 1.7 x 19 x 7.3; weight is 6.8 lbs.
Price: $695.00.

The XEQ-2 is a mono 2-way electronic crossover/equalizer incorporating third-order Butterworth filters for 18 dB/octave slopes. It has low frequency Thiele equalizer, variable time delay, plug-in crossover frequency and high frequency CD horn eq. Dimensions are 1.7 x 19 x 5; weight is 4.75 lbs.
Price: $534.00.

The EX-18 is a stereo 2-way, mono 3-way electronic crossover with variable frequency control from 100 Hz to 16 kHz (with 10 X switch), third-order Butterworth filters for 18 dB/octave slopes, high frequency polarity reverse switch, and actively balanced inputs with low impedance outputs. Dimensions are 1.75 x 19 x 5; weight is 4 lbs.
Price: $341.00.

FOSTEX
The EN3020 is an electronic stereo 2 or 3-way, mono 4-way crossover. It has a 2-way mode sub woofer output, switchable 12 or 18 dB/octave filters, and balanced and unbalanced inputs and outputs. Dimensions are 3.5 x 17 x 8.25.
Price: $699.00.

FURMAN SOUND
The Model TX-3 is a tunable 2-way stereo, 3-way mono electronic crossover. Each crossover point is adjustable over the full audio range. It has 12 dB/octave Butterworth filters, separate input level controls for each channel and separate output level controls for each band. Dimensions are 1.75 x 19 x 8; weight is 7 lbs.
Price: $295.00.

The TX-4 is the same as the TX-3, but it has 3-way stereo and 5 way mono capability. Dimensions are 1.75 x 19 x 8; weight is 8 lbs. Also available in 4-way stereo model.
Price: $444.00; 4-way stereo model: $505.00.

JBL Model 5234A Electronic Frequency Dividing Network
Single rack space package includes relay-deyaled turn-on circuits andvariable high-pass function/ mono-stereo dip switch banks for LF speaker protection or 6th-order alignment of bass enclosures.Plug-in frequency cards available for power response correction of HF horns. Either 12 or 18 dB/octave slopes available. Dimensions: 19 x 1.75 x 7.7; weight: 4 lbs.
Price: $450 (crossover) $27-$33 (plug-in cards).

LT SOUND
The ECU-2 is a stereo electronic crossover unit capable of stereo biamping as well as stereo triamping. Crossover points are continuously variable from 70 Hz to 11 kHz. It has 12 dB/octave Butterworth filters, summed mono output for subwoofer operation, and individual phase inversion switches on mid and high bands. Dimensions are 1.75 x 19 x 7.5; weight is 6.4 lbs.
Price: $295.00.

PEAVEY
The V4X is a variable 4-way electronic crossover with 24 dBV input and output capabilities on all bandpass outputs to eliminate crossover headroom problems. It has third-order, state variable filters, balanced XLR and 1/4-in. inputs and calibrated frequency adjustment. Dimensions are 1.75 x 19 x 9; weight is 8 lbs.
Price: $379.50.

RANE
The Model AC 22 is a state variable time correcting crossover that is stereo 2-way, mono 3-way with 75 Hz to 3.6 kHz crossover range. It has Linkwitz-Riley 24 dB/octave filters with no phase shift or lobing errors, 0 to 2 ms variable time delay for phase alignment of non-coincidental drivers, 41-detent frequency selectors, channel mute switches, and automatic internal switching for mono configurations.
Price: $389.00.

The Model AC 23 is a stereo 3-way, mono 4 or 5-way version of the AC 22. Crossover ranges are 70 Hz to 1 kHz, and 450 Hz to 7 kHz.
Price: $499.00.

ROSS

The CN3201 is a 3-way stereo, 5-way mono active electronic crossover with phase selectable outputs, balanced inputs, variable frequency select, 12 dB/octave filters, 95 dB S/N, and a THD of 0.005%.
Price: $349.95.

UREI Model 525 Electronic Crossover

Front panel mode switch for stereo 2-way or 3-way, or mono 4-way or 5-way operation. 18 dB/octave slopes for unity summing and maximally flat response. Built-in frequency counter measures and displays crossover frequencies with 1 Hz resolution. Continuously variable crossvers possible from 50 Hz to 10 kHz. Dimensions: 19 x 3.5 x 9.75; weight: 10 lbs.
Price: $796.

YAMAHA

The F1030 is a 2 or 3-way electronic crossover. It has 3 high pass and 2 low-pass filters, a 40 Hz, 12 dB/octave high-pass filter, frequency selectable crossover points from 250 Hz to 8 kHz, at 12 or 18 dB/octave, and 26-detent input and output attenuators. Dimensions are 3.75 x 19 x 9.5; weight is 16.5 lbs.
Price: $675.00.

The F1040 is a 3 or 4-way electronic crossover with 3 low-pass, and 3 high-pass filters. Frequency selectable crossover points range from 70 Hz to 8 kHz at 12 or 18 dB/octave. It has 26 detent input and output attenuators. Dimensions are 3.75 x 19 x 12; weight is 17.6 lbs.
Price: $835.00.

EQUALIZERS

ALTEC LANSING

The 1650B is an active eq with 28 cut only filter sections centered at ISO preferred 1/3-octave center frequency from 31.5 Hz to 16 kHz. It has detented slide controls to provide up to 15 dB of cut, 18 dB/octave high-pass and low-pass with continuously variable 3 dB down frequencies. Dimensions are 5.25 x 19 x 8; weight is 17 lbs.
Price: $1,808.00.

The 1651A is a single-channel active eq that has 10 minimum phase shift filter sections centered at ISO preferred 1/3-octave frequencies. Center detent slide controls provide 12 dB of cut or boost. Dimensions are 3.5 x 19 x 11.25; weight is 12 lbs.
Price: $852.00.

The 1652A is a dual-channel version of the 1651A. Weight is 13.25 lbs.
Price: $1,196.00.

APPLIED RESEARCH & TECHNOLOGY

The 171 is a dual-channel, 2/3-octave eq with minimum phase shift active combining filters centered on ISO preferred frequencies. Dimensions are 3.5 x 19 x 6.25; weight is 7 lbs.
Price: $375.00.

The 172 is a 1/3-octave eq with minimum phase shift active combining filters centered on ISO preferred frequencies. Dimensions are 3.5 x 19 x 6.25; weight is 7 lbs.
Price: $400.00.

The 173 is a dual-channel 2/3-octave balanced eq with minimum phase shift active combining filters centered on ISO preferred frequencies. Dimensions are 3.5 x 19 x 6.25; weight is 7 lbs.
Price: $495.00.

The 174 is a 1/3-octave balanced eq with minimum phase shift active combining filters centered on ISO preferred frequencies. Dimensions are 3.5 x 19 x 6.25; weight is 7 lbs.
Price: $495.00.

ARIA MUSIC

The SQ520 is a 10-band stereo eq with spectrum analyzer. Dimensions are 5 x 19 x 12.
Price: $459.00.

The EQ531 is a 31-band, 1/3-octave eq. Dimensions are 3 x 19 x 8.
Price: $349.00.

The EQ515 is a 15-band stereo eq. Dimensions are 3 x 19 x 8.
Price: $339.00.

The EQ522 is a 10-band stereo eq. Dimensions are 3 x 19 x 8.
Price: $239.00.

AUDIO + DESIGN

The E900-RS sweep eq has 4 independent, overlapping sweep frequency bands, 20 dB boost and cut, 2 available "Q" ranges, and fixed bandwidth. It can be operated in dual mono or stereo.
Price: $790.00.

The SCAMP SO3 is a modular 3-band version of the E900-RS.
Price: $325.00.

The SCAMP SO4 parametric eq offers 3 independent fully parametric sections with overlapping coverage of the audio band. The LF section is variable from 20 Hz to 1 kHz, the MF from 75 Hz to 7.4 kHz, and the HF from 400 Hz to 20 kHz. Both the high and low bands may be switched to a shelving characteristic, the slope being governed by the bandwidth control.
Price: $425.00.

The SCAMP SO7 octave eq is a 10-section octave/graphic eq with standard operating frequencies. It has a 24 dB control range.
Price: $280.00.

AUDIO LOGIC

The SC-31 is a 31-band 1/3-octave graphic eq with ISO centered frequencies, 12 dB of boost/cut or cut only, XLR, 1/4-inch, TRS phone and barrier strip connections, an LED level indicator, variable frequency pass and low-pass filters, and high slew rate, low noise operation. Dimensions are 3.5 x 19 x 6.75; weight is 6 lbs.
Price: $499.95.

BIAMP SYSTEMS

The EQ 220 is a dual-channel, 10-band, octave graphic eq with peak indicators, eq bypass, and electronically balanced/unbalanced 1/4-inch phone connections. Dimensions are 3.5 x 19 x 6; weight is 6 lbs.
Price: $329.00.

The EQ-230 is a dual-channel, 15-band, 2/3-octave graphic eq with metering, eq bypass, high-pass filter, and adjustable low pass filter. Dimensions are 3.5 x 19 x 6; weight is 7 lbs.
Price: $529.00.

The EQ-290 is a 29-band, 1/3-octave graphic eq with metering, eq bypass, high-pass filter, adjustable low-pass filter, floating and balanced/unbalanced barrier strip or XLR, 1/4-inch, and RCA connections. Dimensions are 3.5 x 19 x 6; weight is 6.75 lbs.
Price: $549.00.

The EQ-140 is a 4-band parametric eq with peak indicators and eq bypass for each filter and master, notching/shelving capability, floating and balanced/unbalanced XLR and 1/4-in. phone connections, and ground strap. Dimensions are 1.75 x 19 x 6; weight is 5 lbs.
Price: $399.00.

CARVIN

The EQ-2020 is a dual-channel, 10-band graphic eq with accurate summing characteristics. Dimensions are 3.5 x 19 x 7; weight is 11 lbs.
Price: $259.00.

The EQ-2029 is a 1/3-octave graphic eq with precision summing, calibration filters, and balanced inputs and outputs. Dimensions are 3.5 x 19 7; weight is 11 lbs.
Price: $279.00.

dbx

The Model 905 is a 3-band parametric eq with precise shaping of frequency curve, controls for reciprocal boost/cut (up to 15 dB), and infinite notch switch on each band. Dimensions are 5.25 x 1.5 x 9.5.
Price: $379.00.

The 10/20 is a computerized dual-channel, 10-band eq/analyzer with a 25 dB eq control range, pink noise generator, SPL meter, LED display, and 10-memory storage capability that is useful in setting eq curves for 10 different listening positions. Dimensions are 3.5 x 17 x 12; weight is 12 lbs.
Price: $1,200.00.

DOD

The R 430 is a dual-channel, 15-band graphic eq with up to 12 dB of boost and cut, 2/3-octave ISO centered frequency bands, 0 dB detents, LED level indicators, and switchable low cut filters. Dimensions are 1.75 x 19 x 6.75; weight is 5.75 lbs.
Price: $319.95.

The R 431 is a single-channel, 31-band graphic eq with up to 12 dB of boost or cut, 1/3-octave ISO centered frequency bands, 0 dB detents, LED level indicators, switchable low cut filter, and electronic in/out switching. Dimensions are 1.75 x 19 x 6.75; weight is 5.75 lbs.
Price: $319.95.

The R 815B is a single-channel, 15-band graphic eq with up to 12 dB of boost and cut, an LED level indicator, switchable low cut filter, 2/3-octave ISO centered frequency bands, and 0 dB detent. Dimensions are 3.5 x 19 x 6.75; weight is 5.75 lbs.
Price: $199.95.

The R 830B is a dual-channel, 15-band graphic eq with up to 12 dB of boost and cut, LED level indicators, switchable low cut filters, input level control, center detent sliders, and 2/3 octave ISO centered frequency bands. Dimensions are 3.5 x 19 x 6.75; weight is 5.75 lbs.
Price: $299.95.

The R 831B is a graphic eq with 31 1/3-octave ISO centered frequency bands with 12 dB of boost and cut, LED level indicator, switchable low cut filter, electronic in/out switching, and 0 dB detents. Dimensions are 3.5 x 19 x 6.75; weigh is 5.75 lbs.
Price: $329.95.

ELECTRO-VOICE

The EVT 2230 is a 27-band, 1/3-octave graphic eq with ISO center frequencies, 12 dB boost/cut, true combining filter action, switchable high- and low-pass filters, peak LED, actively balanced inputs and outputs, and eq in/out switch. Dimensions are 3.5 x 19 x 7; weight is 12 lbs.
Price: $642.00.

The EVT 2210 is a dual-channel, 10-band graphic eq with ISO center frequencies, 12 dB boost/cut, peak LEDs, eq in/out switch, and gain control for each channel. Dimensions are 3.5 x 19 x 7; weight is 13 lbs.
Price: $470.00.

FOSTEX

The 3030 is a 10-band graphic eq that is rack adaptable. It has unbalanced inputs and outputs. Dimensions are 3.5 x 17 x 8.25.
Price: $250.00.

FURMAN

The PQ-3 is a 3-band parametric eq/preamp with instrument preamplifier, constant "Q" circuitry, extremely narrow notches, and 20 dB eq range. Dimensions are 1.75 x 19 x 8; weight is 7 lbs.
Price: $321.00.

The PQ-6 is a dual-channel version of the PQ-3. Weight is 9 lbs.
Price: $535.00.

The SG-10 is a graphic eq/preamp combining the best features of a parametric and graphic eqs. It is switchable between 5-band stereo and 10-band mono modes. Dimensions are 3.5 x 19 x 8; weight is 8 lbs.
Price: $395.00.

HOSHINO

The GE1500 is a dual-channel graphic eq with individual in/out switches for each channel. It also has a boost/cut range selction switch and switchable high-pass filter.
Price: $350.00.

The GE3100 is a 2/3-octave graphic eq with boost/cut range selection switch, and switchable high-pass filter.
Price: $310.00.

JBL/UREI Model 5547 Graphic and 5549 Room Equalizers

Thirty boost/cut bands. Fully active-custom hybrid filter amplifiers.Unique gain structure controls optimize headroom and signal to noiseperformance with one operation for any equipment interface. High and low frequency tunable end cut filters with bypass. Active and automatic passive (power loss) bypass modes. XL, phone jack and barrier strip terminals. 5549 is identical except 30 band controls are cut-only. Dimensions: 19 x 3.5 x 8; weight: 9.5 lbs.
Price: $798 (5547) $849 (5549).

KLARK-TEKNIK

The DN360 is a dual-channel, 1/3-octave graphic eq. It has 2 discrete channels of 30-band, 1/3-octave equalization utilizing proprietary MELT circuitry. Each channel includes level control, subsonic filter and a scale switch (6 or 12 dB) to control the boost/cut range. Dimensions are 5.25 x 19 x 8; weight is 10 lbs.
Price: $1,625.00.

The DN300 is a single-channel, 1/3-octave boost/cut graphic eq including high and low-pass adjustable filters, 30 bands of control, bypass relay, ground lift switch and electronically balanced inputs. Dimensions are 3.5 x 19 x 8; weight is 8 lbs.
Price: $995.00.

The DN301 is a cut only (15 dB) version of the DN300.
Price: $995.00.

The DN332 is the same as the DN300, except it is dual-channel, 16-band, 2/3-octave boost/cut without band pass filters and bypass relay.
Price: $1,095.00.

LT SOUND

The PEQ-2 is a dual-channel, 4-band parametric eq with selectable peak/dip or shelving response on upper or lower bands, overall hard-wire bypass and individual bypass on middle two bands. The bandwidth is variable from 0.15 octave to 2 octaves. Dimensions are 3.5 x 19 x 7.5; weight is 11 lbs.
Price: $595.00.

The PEQ-1 is a single-channel version of the PEQ-2. Dimensions are 1.75 x 19 x 7.5; weight is 5 lbs.
Price: $349.00.

GOTHAM/NEUMANN

The W492 is a 5-band transformerless 5-filter eq that is continuously variable, with high-pass switchable filter. Dimensions are 2 x 9 x 8; weight is 2 lbs.
Price: $710.00.

The W495B is a 3-band, 3 filter eq with switchable bandwidth, stepped settings, shelving, and 10 dB band filter. Dimensions are 2 x 9 x 8; weight is 2 lbs.
Price: $825.00.

The W495STB is a dual-channel version of the W495B.
Price: $1,640.00.

The NTP 585 is a 4-channel automated eq system. It is a 14-band graphic eq with digital tape backup, memory presets, and 2/3 octave frequency centers. Dimensions are 7 x 19 x 18; weight is 35 lbs.
Price: $14,000.00.

ORBAN

The 622B is a dual-channel parametric eq with continuous non interacting control over center frequency, bandwidth, and amount of boost/cut. It has line-level balanced input and unbalanced output. Dimensions are 3.5 x 19 x 5; weight is 10 lbs.
Price: $879.00.

The 672A is a single-channel quasi-parametric eq with continuous control over center frequency, bandwidth, and amount of peak boost/cut. Graphic-style eq controls provide reciprocal eq in 8 bands. Dimensions are 5.25 x 19 x 5.25; weight is 8 lbs.
Price: $689.00.

The 674A is a 2-channel version of the 672A. Weight is 11 lbs.
Price: $1,299.00.

PEAVEY

The EQ-215 is a dual-channel, 15-band, 2/3-octave eq with selectable range for 6 or 12 dB boost/cut, 40 Hz low, and 16 kHz high cut filters, bypass, level controls, balanced and unbalanced inputs and outputs, and 24 dBV output capability. Dimensions are 1.75 x 19 x 12; weight is 7 lbs.
Price: $359.50.

The EQ-31 is a 31-band, 1/3-octave graphic eq featuring 6 or 12 dB boost/cut, variable high and low cut filters, level control, bypass, balanced and unbalanced inputs and outputs, and 24 dBV output capability. Dimensions are 1.75 x 19 x 10; weight is 6 lbs.
Price: $349.50.

The EQ-27 is a 27-band, 1/3-octave graphic eq featuring level control with 15 dB range, balanced and unbalanced inputs and outputs, and 19 dBV output capability. Dimensions are 5.25 x 19 x 9.25; weight is 18 lbs.
Price: $449.50.

ROLAND

The SEQ-331 is a single-channel, 31-band, 1/3-octave graphic eq with 12 dB control range, level control, and 3 point detented sliders. Dimensions are 3.6 x 19 x 13.8; weight is 9.25 lbs.
Price: $495.00.

The SEQ-315 is a dual-channel, 15-band, 2/3-octave graphic eq with 12 dB control range, 3 point detented sliders, front and rear panel inputs, bypass switch, and input/output level LEDs. Dimensions are 3.6 x 19 x 13.8; weight is 9.25.
Price: $550.00.

The RPQ-10 is a parametric eq/preamp with 2 parametric eq bands. Level control sets the boost/cut level to +/-15 dB. It has 2 LED overload indicators, and LED on each slider control, and phone and RCA jack inputs. Dimensions are 9 x 7 x 1.75; weight is 2 lbs.
Price: $195.00.

The RGE-10 is a 10-band graphic eq with level control range of 12 dB, input and output level controls, RCA and phone jacks, and an LED on all slider controls. Dimensions are 9 x 7 x 1.75; weight is 2 lbs.
Price: $195.00.

The GE-7 is a compact effects pedal graphic eq with 7-bands of eq. Overall control range is +/-15 dB. Dimensions are 2.8 x 2.2 x 5; weight is 1 lb.
Price: $130.00.

ROSS SYSTEMS

The R-12SP is a dual-channel, 12-band, parametric eq with selectable 1/4-octave center frequencies, 12 dB boost/cut, and 100 dB S/N ratio.
Price: $319.95.

The R-31M is a single-channel, 31-band, 1/3-octave graphic eq with XLR and 1/4-in. inputs and outputs, 12 dB boost/cut, input level control, LED peak indicator, and status switch.
Price: $199.95.

The R-15S is a dual-channel, 15-band, 2/3-octave graphic eq with 1/4-in. inputs and outputs, 12 dB boost/cut, input level control, LED peak indicator, and status switch.
Price: $199.95.

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SOUNDCRAFTSMEN

The TG3044 is a dual-channel, 1/3-octave graphic eq with selectable subsonic filters for each channel. It has balanced or unbalanced inputs and outputs. Dimensions are 5.25 x 19 x 11; weight is 18 lbs.
Price: $689.00.

The AE2000P is a dual-channel graphic eq/RTA with built-in pink noise generator, variable-rate automatic or manual octave scanning, and adjustable pink noise level. The dimensions are 5.25 x 19 x 11.25; weight is 24 lbs.
Price: $799.00.

The G2241 is a dual-channel graphic eq with pre/post processor loops, LED indicators, bypass, and standard 1-octave ISO center frequencies. Dimensions are 3.5 x 19 x 9; weight is 13 lbs.
Price: $299.00.

TEAC

The PE-40 is a 4-channel, 4-band parametric eq.

The GE-20B is a dual-channel, 10-band graphic eq with 12 dB boost/cut at 1-octave ISO frequencies. Each channel has a pair of 12 dB/octave cutoff filters.

UREI 533/535

Synthesized LC filter circuitry with smooth combining characteristic.Gain variable from -10 to +20 dB, wide dynamic range. Low distortion, low noise, minimum phase shift. 10 adjustable equalizers on ISO one-octave center frequencies. Model 535 is dual equalizer based on 533.Dimensions: 533 - 8.5 x 3.5 x 8, 535 - 19 x 3.5 x 8; weight: 6.5 lbs, 9.5 lbs.
Price: $366 (533); $496 (535).

UREI 537 Graphic Equalizer

27 calibrated controls. True LC active filter circuits. Low distortion, lowest noise, minimum phase shift. Zero insertion loss, up to 20 dB gain. Signal-to-noise ratio over 110 dB. Dimensions: 19 x 3.5 x 8; weight: 11 lbs.
Price: $896.

UREI 539 Room Equalizer. 27 calibrated controls. True LC active filter circuits. Low distortion, lowest noise, minimum phase shift. Zero insertion loss, up to 20 dB gain. Signal-to-noise ratio over 110 dB. Dimensions: 19 x 3.5 x 8; weight: 11 lbs.
Price: $1046.

UREI 546 DUAL PARAMETRIC EQUALIZER

Most flexible EQ filter set. Two independent channels, may be switched to expanded single channel mode, providing 8 parametric filters. All adjacent filters have overlapping ranges. Bypass switches for each channel and each individual filter. Reciprocal response curve in boost and cut. Tunable wide-range end cut filters. Gain variable over 30 dB range. Dimensions: 19 x 5.25 x 8; Weight: 13 lbs.
Price $746.

UREI 565T "Little Dipper" Filter Set. Solves problems with unwanted noise in the audio band. Rejects coherent and semi-coherent signals from program material. Notch filters continuously tunable (20 Hz - 20 kHz) with selectable bandwidth. Program content not affected in extreme narrow notch position. Variable sharp low and high cut filters to reduce noise at band ends. Dimensions: 19 x 5.25 x 9; Weight: 13 lbs.
Price: $1196.

WASHBURN

The WEQ-31M is a single-channel, 31-band, 1/3-octave graphic eq with peak overload LED and a 20 Hz to 20 kHz bandwidth. Dimensions are 1 x 19 x 8; weight is 3 lbs.
Price: $349.00.

The WEQ-15S is a dual-channel, 15-band graphic eq with 12 dB boost/cut capability, LEDs indicating peak/overload conditions, bypass capabilities, and effects level controls. Dimensions are 1 x 19 x 3; weight is 3 lbs.
Price: $349.00.

YAMAHA

The Q1027 is a single-channel, 27-band, 1/3-octave eq. It has up to 12 dB boost/cut, 18 dB/octave high-pass at 40 or 80 Hz, transformer isolation in and out, and 24 dB maximum output level. Dimensions are 3.5 x 19 x 12; weight is 17.6 lbs.
Price: $995.00.

The Q2031 is a dual-channel, 31-band, 1/3-octave eq. It has 6/12 dB boost/cut, 12 dB/octave high-pass filter (20-200 Hz), electronically balanced in and out, and 24 dB maximum voltage gain. Dimensions are 3.5 x 19 x 11.75; weight is 11.2 lbs.
Price: $595.00.

The GQ1031 is a single-channel, 31-band, 1/3-octave eq. It has up to 12 dB boost/cut, an unbalanced input and 600 ohm output, and 20 dB maximum output level. Dimensions are 1.75 x 19 x 8.75; weight is 6.4 lbs.
Price: $265.00.

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Audio/Design
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Bremerton, WA 98310

Audio Logic
see DOD

Biamp Systems, Inc.
PO Box 2160
Portland, OR 97208

Brooke Siren
see Klark Teknik

Carvin
1155 Industrial Ave.
Escondido, CA 92025

dbx
71 Chapel St.
Newton, MA 02195

DeltaLab
One Progress Way
Wilmington, MA 01887

DOD
5639 South Riley Lane
Salt Lake City, UT 84107

Electro Voice
600 Cecil St.
Buchanan, MI 49107

Eventide, Inc.
One Alsan Way
Little Ferry, NJ 07643

Fender Musical Instruments
1130 Columbia St.
Brea, CA 92621

Fostex Corporation of America
14531 Blackburn Ave.
Norwalk, CA 90650

Furman Sound
30 Rich St.
Greenbrae, CA 94904

Gotham Audio Corporation
1790 Broadway
New York, NY 10019

Hoshino (USA), Inc.
PO Box 886
Bensalem, PA 19020

JBL/UREI
8500 Balboa Blvd.
Northridge, CA 91329

Klark Teknik Electronics, Inc.
30 B Banfi Plaza N.
Farmingdale, NY 11735

Lexicon, Inc.
60 Turner St.
Waltham, MA 02154

LT Sound
PO Box 338
Stone Mountain, GA 30086

Morley
6855 Vineland Ave.
North Hollywood, CA 91605

Orban Associates
645 Bryant St.
San Francisco, CA 94107

Peavey Electronics
711 'A' St.
Meridian, MS 39301

Rane Corp.
6510 216th SW
Mountlake Terrace, WA 98043

RolandCorp US
7200 Dominion Circle
Los Angeles, CA 90040

Ross
PO Box 2344
Fort Worth, TX 76113

Soundcraftsmen, Inc.
2200 So. Ritchey
Santa Ana, CA 92705

Teac Corporation of America
7733 Telegraph Rd.
Montebello, CA 90640

Washburn International
230 Lexington Dr.
Buffalo Grove, IL 60090

Yamaha International Corp.
6600 Orangethorpe Ave.
Buena Park, CA 90620

People, Places...

• **ARS Recording Studios** in Alsip, IL, has recently installed a keyboard section in its control room consisting of: Yamaha DX-7, TX-7, Ensoniq Mirage, Casio CZ 101, Roland Planet-S, 360 Systems MIDI Bass, Linn Sequencer, 32-track MIDI recorder and Linn Drum drum machine.

• **Paul Gordon** has been appointed manager, advertising for Konica Professional and Consumer Products Division, **Konica, USA, Inc**. He will be responsible for national consumer and trade advertising as well as coordinating production of point of purchase support material for the dealers. Gordon joined Konica, USA, Inc, after four years as a technical representative for Minolta's Northeast region. Konica USA, Inc, is involved in all aspects of the imaging industry, including 35mm cameras and accessories, color print film, mini-labs, etc.

• **Biamp Systems** is pleased to announce the appointment of three new territory representatives. They are: **Dave Spalding** of Double Edge Marketing, covering the states of Maine, New Hampshire, Vermont, Massachusetts, Connecticut, and Rhode Island; **Rick Parent** and **Kevin Talment** of Fleetwood Marketing, covering the states of Wisconsin, Michigan, Illinois, Indiana, and Kentucky; and **Pete Wood** of Woodco, covering the states of Texas, Oklahoma, Arkansas, Mississippi, West Tennessee, and Louisiana.

• **Michael Cresci** has joined **Clarion Corporation of America** as product manager. In this position, Cresci will be responsible for new product concepts and planning for Clarion as well as field sales training. Most recently Cresci was technical applications engineer for Sony Corporation's Autosound Division. At Sony, he worked on new products and travelled throughout the country conducting sales training sessions for installers.

• **Walter W. Wurfel** will join **NAB** as senior vice president, Public Affairs and Communications Department. He succeeds Shaun Sheehan who is now. vice president, Washington, Tribune Broadcasting Co. Wurfel is president of Ruder Finn & Rotman/Washington, a New York-based public relations firm. Previously, he was vice president, Corporate Communications, Gannett Co. from 1979 to 1984. He served as President Carter's deputy press secretary for two years, and was field press director of the 1976 Carter-Mondale campaign. He also worked on Capitol Hill as press secretary to US Senator Richard Stone. From 1965 to 1966 he was assistant to the president, Straus Broadcasting Group, a New York radio group and prior to that was assistant news director and anchorman for WTSJ-TV in San Juan, Puerto Rico. Wurfel also served as a reporter and editor for newspapers and newsletters. NAB serves a membership of over 4,600 radio and 900 television stations.

• **TekCom,** one of Philadelphia's professional audio dealers, has recently completed two audio installation including an upgrade of the house sound system in the world-renowned Academy of Music and a sound reinforcement system for the new cabaret in the Trump Castle in Atlantic City, NJ. Both systems are based on EAW loudspeaker products.

• The professional products division of **Bose** Corporation has announced two marketing appointments: **Barry R. Luz** as product planner and **Mark R. Mayfield** as marketing development specialist. Formerly the company's field sales representative for the midwest states, Luz will now be responsible for coordinating product development activites for new professional products from Bose. As marketing development specialist, Mayfield will be involved in preparing new marketing programs and analyzing the potential of new market segments.

• **John T. Hartley** has been elected president and chief executive officer of Harris Corporation. Hartley was president and chief operating officer and succeeds Dr. Joseph A. Boyd who remains chairman of the board. Hartley was elected president of Harris in 1978. He joined the company as research engineer in 1956 and has held positions of increasing management responsibility during his thirty year career with Harris. As Harris president and chief operating officer, Hartley's recent focus has been on streamlining operations to increase profitability, with particular emphasis on lower product costs, reduced expenses, new product introductions, and increased marketing effectiveness. Harris Corporation produces state-of-the-art information processing, communication and microelectronic products for the worldwide information technology market. The company operates thirty-five plants in the US and abroad.

• **City Sound Recording** of San Francisco, CA, has added new equipment: an ADR pan scan auto panner, a DBX 165 limiter, and a fully updated AMS RMX 16 reverb.

• **Star Track Recording Studios**, a commercial music studio based in Tulsa, OK, has changed its name to **Universal Music & Post, Inc.**, as part of the company's expanded direction. Star Track Recording was founded in 1978.

• **Gary S. Post** has been appointed as product specialist for the Design Acoustics division of **Audio-Technica**. His duties will include sales training and administrative processing of Design Acoustic dealer inquiries for both home and mobile loudspeaker systems. Formerly a store manager of Ohio Sound, a Design Acoustics dealer, Post brings to the assignment a good working knowledge of the line and familiarity with the retail atmosphere in which loudspeaker systems are sold.

& HAPPENINGS

JOINT VENTURE FOR COMPACT DISC-RELATED PROFESSIONAL STUDIO SYSTEMS

Willy Studer A.G. and **N.V. Philips** intend to form a joint venture on a 50/50 basis for the research and the development of Compact Disc-related professional studio systems. The joint venture is intended to exploit the synergies of the two companies in R and D resources, and know-how in product engineering in product engineering. The parties expect to optimize their marketing efforts both in product range and in distribution channels. The venture will not affect ongoing independent developments by both companies in the areas of magnetic tape recording and optical disc mastering systems.

AKG ESTABLISHES DIGITAL PRODUCTS DIVISION

AKG Acoustics, Inc., has announced the acquisition of **Ursa Major**. As of April 1986, AKG acquired all assets and trademarks of Ursa Major, thus formally establishing a new "digital products division" within AKG Acoustics. In addition to the extensive R&D activities undertaken by the parent company, AKG Acoustics, Vienna, Austria. The new Boston-based facility will become AKG's second R&D center for digital product development. AKG Acoustics' Stamford facility will handle all US sales and marketing, export, and administration of the new division. Christopher Moore, newly appointed executive vice president of the division, will be in charge of all future projects and will head up the product development team utilizing the talents of the current staff. Two new products introduced under the AKG name at the recent NAB show in DAllas, TX, represent an advanced use of the digital technology available today as applied to reverberation/effects and stereo processing. AKG is a manufacturer of microphones, headphones, reverb and delay systems, phonocartridges, and related audio products.

NAB/NRBA RADIO '86 COMMITTEE ANNOUNCED

The **National Association of Broadcasters** has announced the committee members for the **Radio '86 Convention**. The third annual meeting jointly sponsored by NAB and NRBA (National Radio Broadcasters Association) will be held September 10-13 at the New Orleans Convention Center and the New Orleans Marriott. Co-chairmen are: NAB Radio Board Chairman John F. Dille III, president, Federated Media, Elkhart, IN, and NRBA Director at Large, Joe Dorton, president, Gannett, Radio Division, St. Louis, MO.

COMPUSONICS FORMALLY INTRODUCES DSP-1500

CompuSonics Corporation formally introduced its DSP-1500 digital audio system at the 1986 NAB show in Dallas, TX. The DSP-1500 is a floppy disc-based digital recording and playback system that is designed to be a plug-in replacement for the traditional broadcast cart machine used in radio stations across the country.

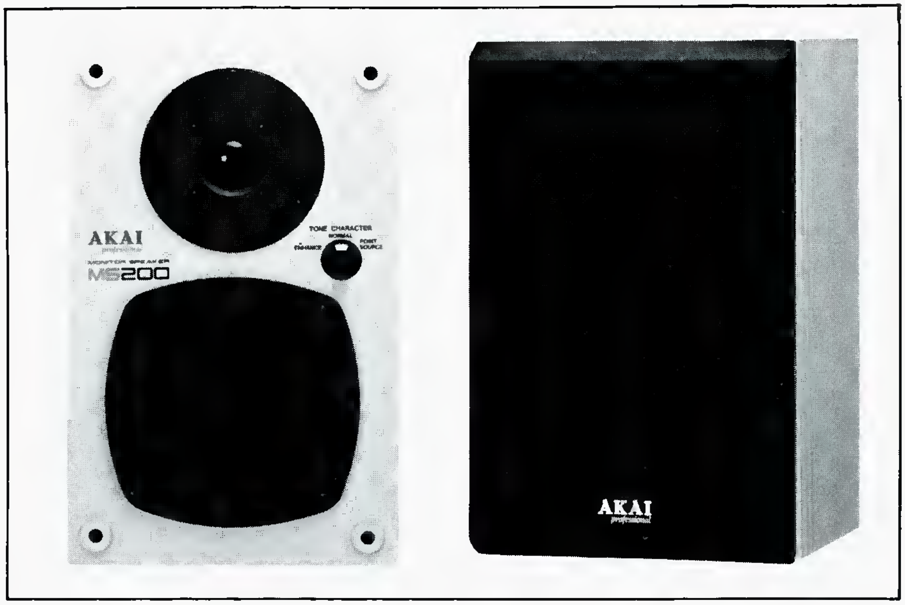
BERKLEE HONORED BY YAMAHA INTERNATIONAL

Berklee College of Music has been honored as the first institutional recipient of the Yamaha Music Award, which is presented annually by the Yamaha International Corporation in recognition of "outstanding contribution to the popularization of music and inspiration to musicians worldwide." In accepting the award from Yamaha at a recent ceremony in Anaheim, CA, Berklee president Lee Eliot Berk noted Yamaha's longstanding achievements in music education and contributions to the technologies which have become so much an essential part of today's professional music. Also receiving 1986 Yamaha Music Awards were singer/songwriter Lionel Richie, record producer Phil Ramone, FM synthesis pioneer John Chowning, film music composer Jerry Goldsmith, singer/songwriter Joni Mitchell, keyboardist/composer Chick Corea, bassist Nathan East, and keyboardist/vocalist Donald Fagen. Berklee College of Music is renowned for its illustrious jazz faculty and practical career preparation.

New Products

AKAI MONITOR

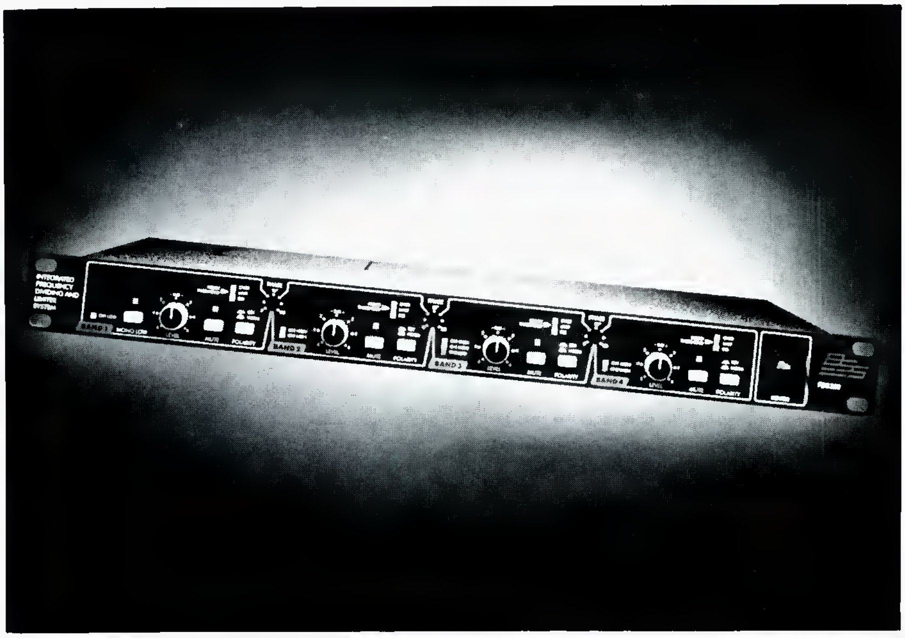
• The Akai Professional Products MS 200 Multimonitor has been designed to eliminate the barrage of monitors usually used in the recording studio. It incorporates three typical reference monitor equalization curves in one box: a single source point full range 4.5-in. loudspeaker monitor for perfect imaging, a flat response 2-way monitor flat from 60 Hz to 23 kHz for balance and fidelity, and an enhanced setting for the typical standard studio monitor control room sound, all selectable with a front panel 3-way switch. The MS 200 features a 4.5-in. full range woofer with a high powered 1-in. soft dome tweeter. It has maximized the power handling capacity of the system with an 80 watt RMS rating and a 200-watt maximum allowing the engineer to achieve the sonic levels desired without fear of damaging the monitors. The frequency response of the MS 200 is 60 Hz-23 kHz, +/-2 dB in flat or enhanced mode. In source point mode, the frequency response is 60 Hz-15 kHz, +/-3 dB. The sensitivity rating of the monitor is 89 dB at 1 watt/1 meter. It comes equipped with a detachable front grill and uses push-on connectors.



Mfr: Akai Professional Products.
Price: $369.95 per pair.
Circle 50 on Reader Service Card

BROOKE SIREN DIVIDER/LIMITER

• The Brooke Siren Systems FDS360 is an Integrated Frequency Divider and Limiter System. It can operate as either a 2-way stereo or 3- or 4-way mono crossover. The front panel includes separate level and mute controls for each of the four sections, and an LED display inidicating signal present, limiter threshold, and over limit conditions. One of the features of the FDS360 is the inclusion of two facilities for correcting signal misalignment due to speaker placement. For adjusting signal phase in the critical crossover region, a 0-180 degree phase control and a polarity reverse switch provide a fairly inexpensive method of reducing the notch filter effects of signal misalignment between bands. For true time correction, a rear panel barrier strip provides patching points into each of the four sections. This allows the insertion of a frequency independent delayed signal into each section from a suitable high quality digital delay. Each section icludes an integral mid-filter limiter for amplifier and speaker protection. The attack and release times for the limiter are determined by the appropriate plug-in frequency card and as such, are optimized for each section. Limiter threshold can be adjusted in 1/2 dB steps via the precision switch network on the rear panel. No external test equipment is necessary.



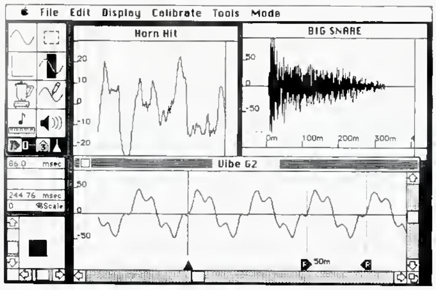
Mfr: Brooke Siren Systems
Price: $1,025,00.
Circle 51 on Reader Service Card

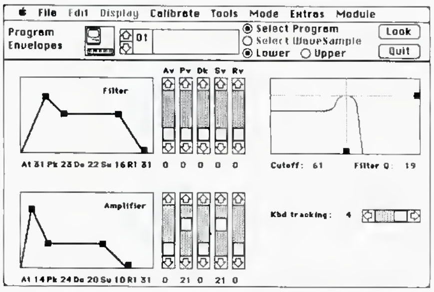
MIRAGE SOFTWARE

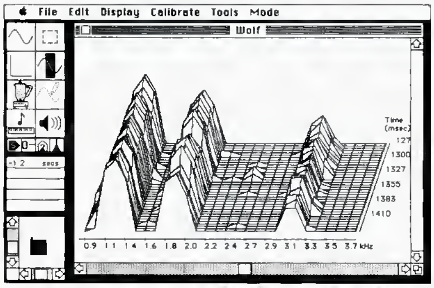
● Digidesign's Sound Designer software is an advanced computer music system for the Apple Macintosh and Ensoniq Mirage digital sampling keyboard. Other versions of Sound Designer are available for the Emu System's Emulator II, and the Sequential Prophet 2000. The software allows sampled sounds to be transferred between the Mirage and the Macintosh using any standard Macintosh MIDI interface. Mirage samples can be stored on Macintosh disks and displayed on the Macintosh's high resolution screen (up to 3 wave-forms can be displayed simultaneously). Extensive sound editing and processing capabilities are provided, including cut and paste editing, waveform drawing, digital mising, digital merging and more, with an editing accuracy of better than 1/50,000 of a second. Waveforms can be scaled to show any degree of detail, from the entire sound to just a few samples. Sound Designer offers many looping aids, including visual looping, cross-fade looping and a loop window that displays the actual "splice" between the beginning and end loop points. An onscreen MIDI "keyboard" (with a built-in track sequencer) can be used to "play" the rack mount Mirage Multi-sampler. Sound Designer also includes advanced digital signal processing functions including FFT (Fast Fourier Transform) based frequency analysis, flexible digital equalization and direct synthesis (sounds are created using synthesis programs on the Macintosh, then transferred to the Mirage for playback). Sound Designer's Front Panel mode provides graphic programming screens that greatly simplify programming of all Mirage parameters and functions. Filter response curves and ADSR envelopes are displayed graphically, and can be drawn using the Macintosh mouse. Keyboard setups, MIDI assignments, sampling parameters, etc., can be quickly adjusted using onscreen menus.

Mfr: Digidesign, Inc.
Price: $395.00.

Circle 52 on Reader Service Card





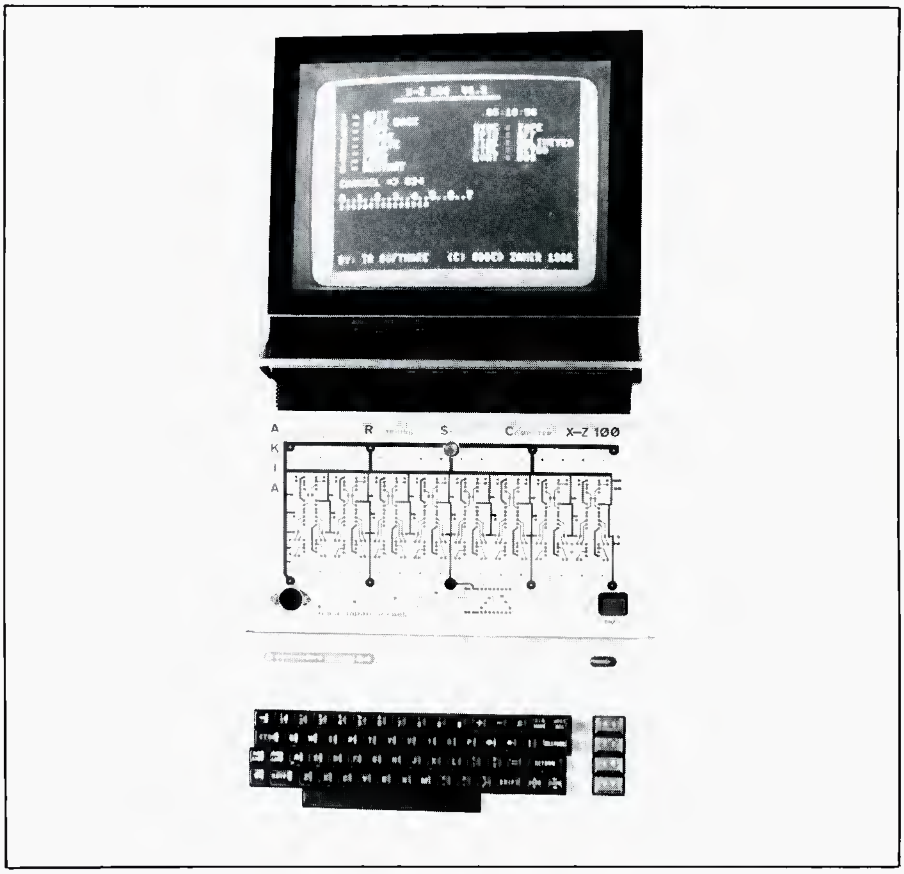


X-Z100 COMPUTER

● The X-Z 100 system can fully automate almost any console, large or small in about half an hour. All the faders will be automated, and since the system can automatically control 128 channels, other parameters of the console, depending on their ease of physical access, can also be controlled. The system works by recording "Time Pointers" on the tape for fully automated gain of each channel, or will work with SMPTE code. Music can therefore be edited in one piece from beginnig to end, or in any segments of any length. When finished, the system will tailor the segments into the final piece. All detailing is easy due to menus on the computer screen. An additional feature of the system is that it acts as a noise gate on any portion of any channel you want. Any portion of any work can be saved to the internal disk and then reloaded into the computer's memory at any time.

Mfr: Akia
Price: $1499.00

Circle 53 on Reader Service Card

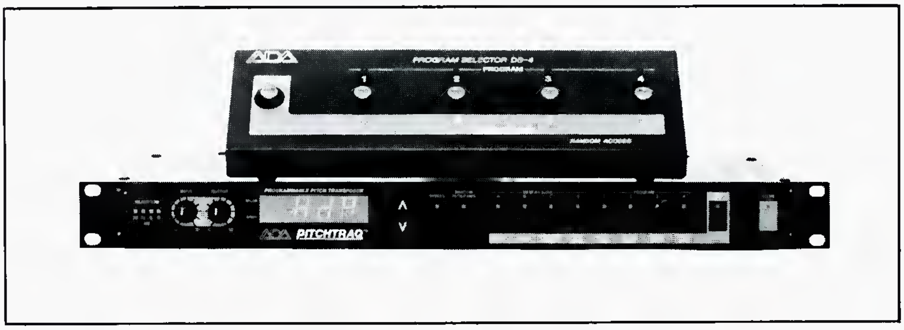


WIREWORKS DESIGN KIT

● The Wireworks Audio Cable Design Kit includes a system specification work sheet and peel-off system components sheet, which provides an overview of the Wireworks Mix and Match Components Group. The Components Group is a multipin-based audio cabling system, comprising over 1000 standard items. The Mix and Match Components allow the user to build an audio cabling system exactly as needed. As you arrange the peel-off Components on the work sheet, the user can visualize their needs in an audio cabling system.
Mfr: Wireworks.

Circle 54 on Reader Service Card

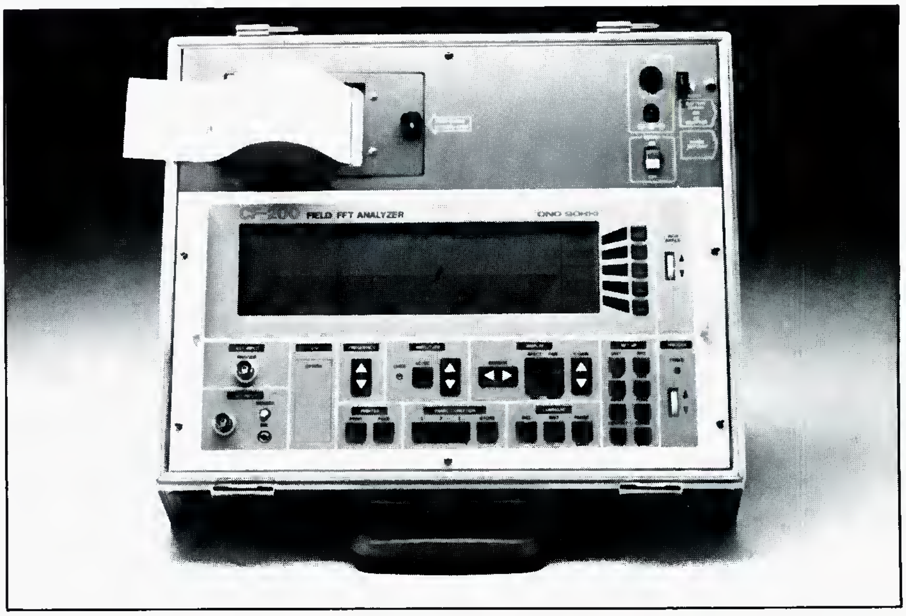
ADA PROGRAMMABLE PITCH TRANSPOSER



● ADA Signal Processors Inc.'s Pitchtraq is a studio quality pitch transposer featuring complete programmability and instant access to any program. The Pitchtraq produces all harmonizing effects within a two octave range, including harmony lines, octave shifts, synthesized textures, de-tuned chorusing, and harmonic alteration. An on-board computer allows full programming of 16 effects, including sweeps, mix, regeneration, and pitch change. In addition, ADA has loaded 16 "shadow" programs into constant memory which may be recalled at any time, or used right out of the box. The optional Footswitch Controller provides remote access to all 16 effects and bypass control. Any effect may be accessed instantly for complete on-stage control. Other features include 15 kHz frequency response, an LED readout which displays pitch change in cents, ratio, or standard musical interval, and a self-diagnostic program which checks the unit during power-up.
Mfr: ADA Signal Processors, Inc.
Price: $599.95.

Circle 55 on Reader Service Card

FFT ANALYZER



● The Ono Sokki CF-200 is a full function, single channel FFT analyzer which provides 400-line and 1/3-octave spectrum analysis over the frequency range of dc to 20 kHz. Processing functions include transient recording and analysis, power and linear spectrum analysis, time-domain averaging, amplitude/frequency listing, and waterfall display. Menu-driven operation and powerful softkey programming make the CF-200 easy to set up and operate. Operation is further expedited by preprogramming up to three panel setups for instantaneous recall in the field. Built-in features include battery power, signal conditioning, and graphics display and printout. Mass memory and computer interface are optional. The unit weighs 20 lbs. and is housed in a rugged carry-on case. Its small size and light weight permit measurement wherever an operator can go. Applications include noise and vibration investigation, machine condition monitoring and trend analysis, and similar measurements in other locations where traditional FFT analyzers are inpractical and impossible to use. Applications also include the testing and qualification of sound reinforcement systems and sound isolation installations.
Mfr: Ono Sokki Co., LTD.
Price: $6,000.00.

Circle 56 on Reader Service Card

Classified

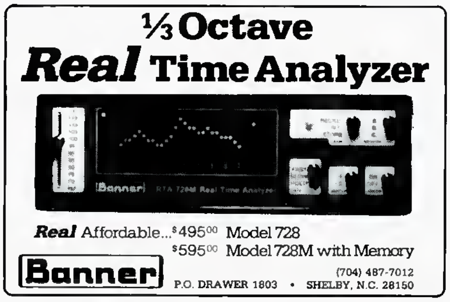
FOR SALE

HEATSHRINK available by the foot in black sizes 3/64" to 1". For more info. and prices write to **Quality Audio Products PO Box 2595 Fairlawn, NJ 07410.**

DAN ALEXANDER BUYS AND SELLS USED AUDIO GEAR, CONSOLES, MICROPHONES. **DAN ALEXANDER AUDIO, PO BOX 9830, BERKELEY, CA 94709. (415) 527-1411.**

RADIO TRANSCRIPTION DISCS: Any size, speed. Drama, comedy, music, variety, adventure, soaps, children's, AFRS, big band remotes, library services. **Kiner-*db*, Box 724, Redmond, WA, 98073-0724.**

The Drum Machine Book. 500 patterns and fills, song arrangements, programming tips, and more—$20.00....128 excellent new **Casio CZ Patches** on data sheets or cassette (with application materials)—$25.00....Casio **CZ Newsletter.** Published quarterly—$10.0....**60** new programs for **Yamaha's REV 7** digital reverb—$20.00....Everything promptly shipped. **Bellweather Records**, Box 22409, Minneapolis, MN 55422. (Ten years in the music business!!)



Closing date is the first of the second month preceding the date of issue.

Rates are $1.00 per word with $25.00 minimum. Boxed ads are $40.00 per column inch. db Box Numbers are $8.50 for wording "Dept. xx" plus $1.50 for postage and handling.

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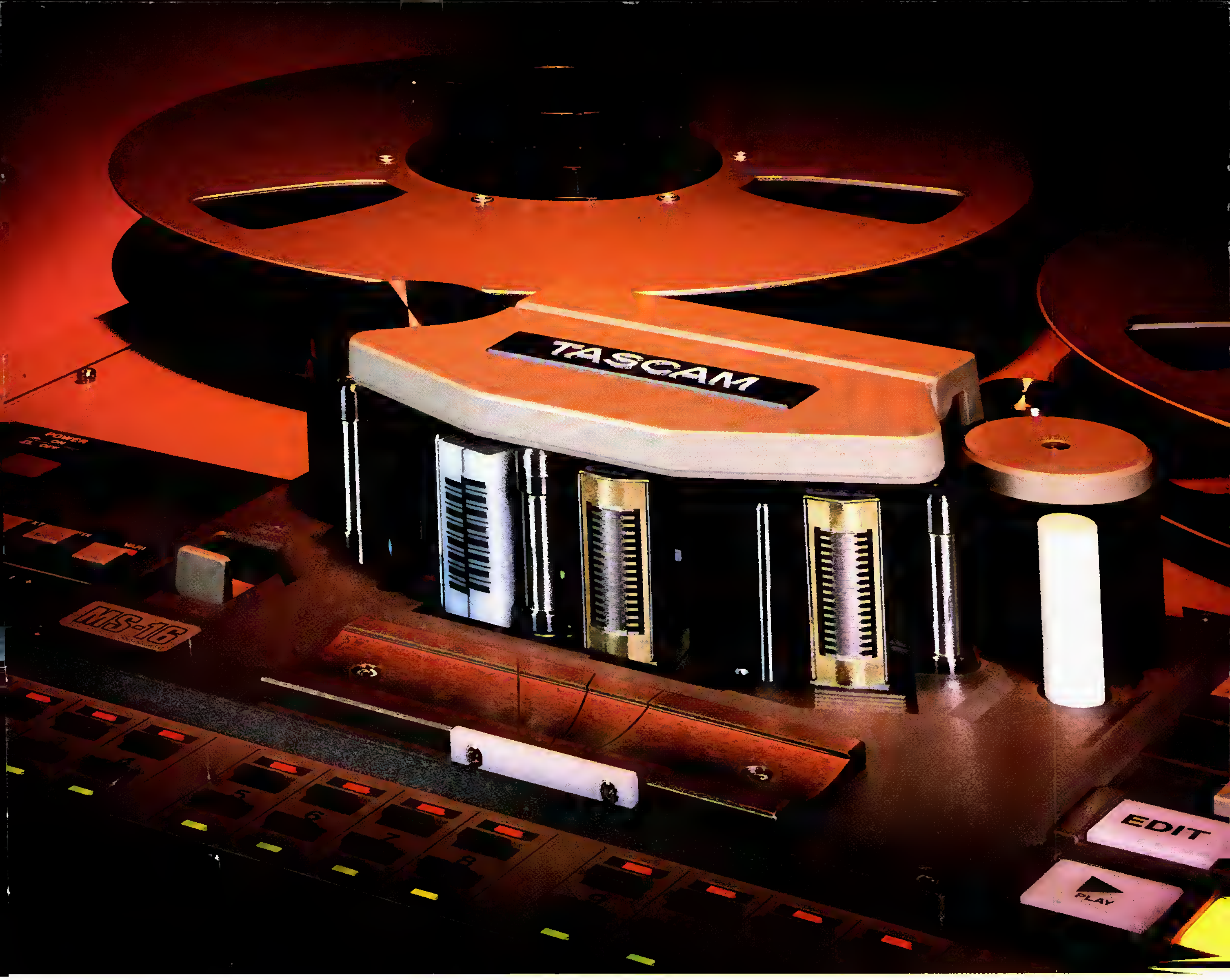


FINISH UP ON TIME WITHOUT SACRIFICING QUALITY.

You want it quick and you want it good. In today's competitive post-production audio/visual scene, the rewards go to those who can produce results that are quick *and* good. That's why TASCAM designed the MS-16 1″ 16-track recorder—to bring together top-notch audio quality plus premium features that streamline production and move you ahead of schedule.

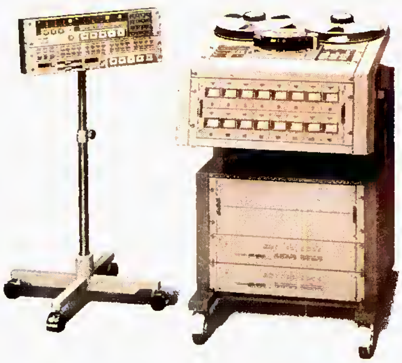
Quality reproduction starts with the heads, and TASCAM has three decades of design experience behind the MS-16's new micro-radii heads. They bring "head bumps" under control and ensure flat frequency response. And unlike most tape machines, the MS-16 record/sync and playback heads are identical in performance. Because sync response equals repro response on the MS-16, you can make critical EQ and processing decisions on overdubs or punch-ins without having to go back and listen a second time. You get what you want sooner and with fewer headaches.

The MS-16 cuts down on the time you spend locking up with other audio and video machines as well. A 38-pin standard SMPTE/EBU interface affords speedy, single-cable connection with most popular synchronizers and editing systems. It's the easy, efficient way to get the most out of today's sophisticated synchronization equipment. The MS-16's new Omega Drive transport is tough enough to stand up to long days of constant shuttling... while handling tapes with the kid-glove kindness they deserve.

Record/Function switches for each track allow effortless, one-button punch-ins. Input Enable allows instant talkback during rewinds, fast forwards and cue searches. These features speed you through sessions and let you concentrate on the project at hand... not on your tape machine.

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

THE TASCAM MS-16 SIXTEEN TRACK



TASCAM THE SCIENCE OF BRINGING ART TO LIFE.

Circle 11 on Reader Service Card

Even if your music starts as a piece of junk, your sampling mic better not.

The new Shure SM94 Condenser Mic can make a big improvement in your digital sampling–at a surprisingly affordable price.

If you've made a major investment in a sampling keyboard or drum machine, don't overlook the importance of the microphone you're using. A vocal mic, for example, might "color" instruments you are sampling.

To capture your sample as accurately as possible, we suggest the new SM94. Unlike many popular mics, the SM94 has no high-frequency peaks, accentuated presence boost, or excessive low-end rolloff. This prevents overemphasis of high frequencies on instruments like strings and brass, while allowing you to retain the important low-frequency response essential to capturing the fullness and richness of many live sounds.

And its extremely low handling noise minimizes the introduction of extraneous handling sounds that might otherwise creep into your sample. What's more, the SM94 offers exceptionally high SPL capability–up to 141 dB–all but eliminating distortion on transient peaks.

For convenience, you can power the SM94 with a standard 1.5 volt AA battery, or run it off phantom power from your mixing board.

In addition to offering a unique combination of features not normally found in condenser mics in its price range, the SM94 is built with Shure's legendary emphasis on ruggedness and reliability. Features like a protective steel case, machined grille and tri-point shock mount make it rugged enough to go wherever your inspiration takes you.

And for voice sampling, we suggest the new SM96 with its vocal contoured response and built-in three-stage pop filter. Both these fine microphones can bring a new dimension of realism to your digital sampling.

For more information, write or call: Shure Brothers Inc., 222 Hartrey Avenue, Evanston, IL 60202-3696, (312) 866-2553.