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THE SOUND ENGINEERING MAGAZINE

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Letters

A TOUCHY SUBJECT

To The Editor:

Norman Crowhurst certainly has a different perspective on copyright protection (see "In My Opinion," Oct. 83—Ed.). He also reveals some interesting prejudices. He apparently has experience as a design engineer and as a writer, but he lacks the perspective of the publisher and manufacturer. To Crowhurst, it's OK to steal as long as you give credit to the author. That may comfort his ego, but it does nothing for the company that put money into development of his work.

I'm intrigued by his references to "inordinately large royalties" and "an extortion racket" based on copyright protection. Anyone who knows the publishing business (be it books, records, or software) knows the fallacy here. The public taste is very difficult to predict, and publishing suffers from a poor hit-to-clinker ratio. Crowhurst grudgingly acknowledges that "a lot of work goes into mastering, working up ideas, and all of that," but his language shows that he really doesn't grasp the amount of investment that a hit product must recoup. It sounds to me as though Mr. Crowhurst has a large chip on his shoulder about the cost of published material.

Unfortunately for publishers, technology is taking away the mechanical barriers to theft of their wares. With Xeroxes in every library and post

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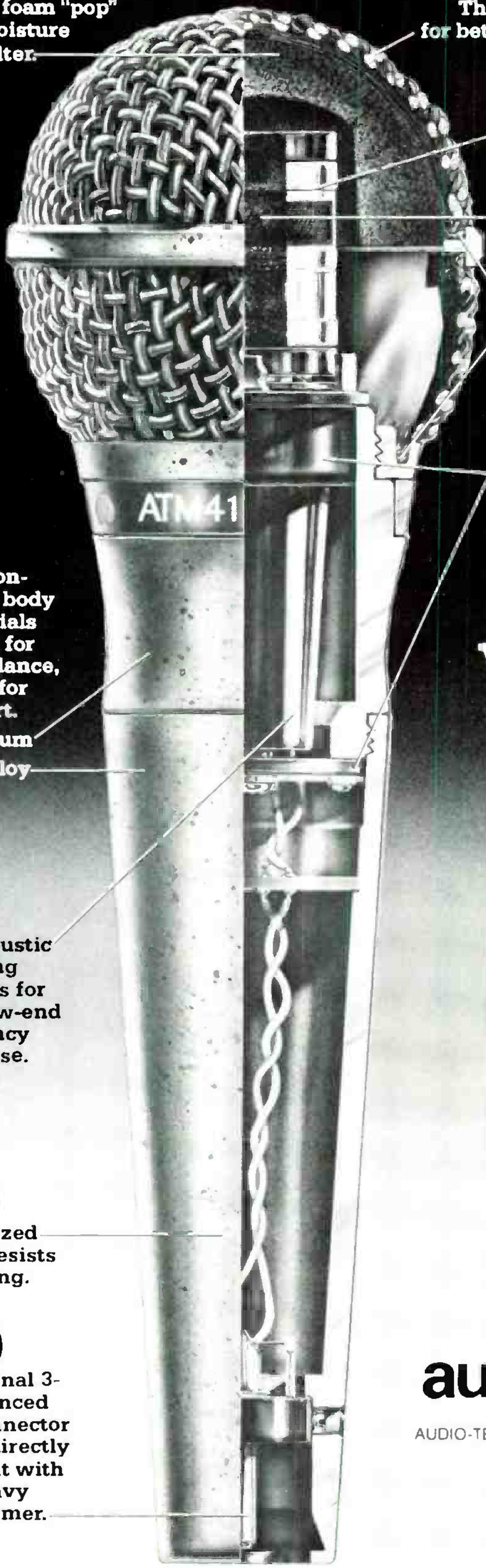
About The Cover

ON THE COVER

• This month's cover features the control room of Dallas Sound Lab, a 48-track facility located in the multi-million dollar Dallas Communications Complex. For the inside story on this outstanding facility, see Martha L. Fischer's article on page 20.



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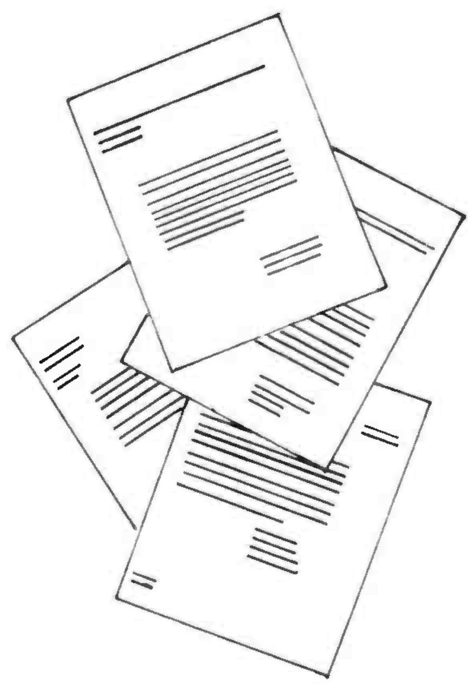
office, and twin-deck portable cassette systems advertised to "beat the system, copy tapes from your friends," legislation won't help protect publishers. Illegal and immoral copying will take place, and the only answer will be to raise the cost of the original release even more.

The point of my April letter was a question: how do we, as professionals in the recording and duplicating business, handle requests for duplicating of copyrighted material? Moral suasion won't work against the customers who copy books or tapes that they have bought or borrowed, but we have a moral interest also. Do we make the copies, and help someone steal, or do we refuse and lose the work to another studio?

HOWARD RUSSELL

There are those questions that we cannot easily answer. There is no question but that a studio making a copy of copyrighted material for a friend or client is doing something that violates the material's copyright. However, the moral aspect of it is something for which we have no answer—except to say that morals and legalities should *(ideally) go together. But can we tell reader Russell that taking on a job involving this type of duplicating is* wrong? *What have those of you faced with similar situations done? We'd be interested in hearing from you.*



Theory & Practice

KEN POHLMANN

This Is Not A Test (Report)

By **Helmut Banger**

• **Nov. 1, 1983.** I am very worried about this test report. How am I supposed to be objective about such a thing? There is more variation in monitor response than anything else in the recording studio, even microphones. Yet monitors dictate all of our actions in the studio because it is only through them that we access the waveforms. Which monitors does one choose? By what right do I decide that a certain pair is best—and thus encode my mix through them, implicitly demanding that anyone else listening to the recording decode the mix through an identical pair. And all of that is contingent on the monitor's interfacing with the room's acoustics. I mean, isn't the room a part of the monitor system too? A pair of monitors that are satisfactory in one room may very well prove to be intolerable in another. Am I thus expected to extend my artistic dictatorship to demand that the listener construct and treat his room similarly, placing his monitors above a window, sixty degrees apart, with his knees tucked under a long reflecting surface covered with little knobs...? Monitors, the most influential link in the recording chain, remain enigmatic to me. When I mix on a pair of trusted monitors, in a trusted room, I feel secure about what I am hearing, but insecure about what it will sound like anywhere else. When I'm in an untrustworthy room, confronted by untrustworthy monitors, I am doubly insecure. I usually close my eyes and pretend I'm back in my trustworthy room. Mentally I equalize and align what I am hearing and try to compensate. When the session is over, I usually have a bad headache.

It is very frustrating for an engineer to work on a pair of arbitrary monitors, and it probably represents the essential challenge of his art to achieve a mixdown that sounds similarly well-balanced through other monitors, in other rooms, or in cars, or at the beach, or through headphones....I guess I'm a coward in that respect; I usually have a pair of trusty monitors in the room, but relatively vary my balance while consulting an assortment of other monitors, both "professional" and "consumer." In addition, I usually run off a cassette and take it home with me for still another opinion. I still remember some AM mixes we were working on—the technician installed an AM transmitter in the rack and we would all run outside and hop in our cars and tune ourselves in on the radio; some of the guys even started up their engines and drove around the parking lot—listening intently and narrowly missing each other. We nicknamed ourselves the compression drivers. The more I think about this, the more depressing it is. Why did I let the magazine talk me into this?

• **Nov. 3, 1983.** I've been thumbing through a lot of engineering texts and monitor design literature for the past two days. I am overwhelmed by their elaborate equations and impressed by the geometrical impact of all those polar plots. The designers make it seem so simple. Textbooks on acoustics even list the qualifications of an ideal monitor:

1. *Electroacoustic efficiency of 100 percent.*
2. *Flat frequency response over audible range.*
3. *Zero harmonic or intermodulation distortion over audible range.*
4. *Faithful reproduction of transients as well as steady-state input signals.*
5. *Nondirectional radiation patterns.*
6. *As small sized as possible.*

I am unable to find much consolation in their engineering confidence. Their equations didn't help me much this morning when my client came back, mad as hell. After paying cash last night, he took a tape over to his girlfriend's house and played it on her solar-powered cassette player... and became convinced that I had defrauded him. How was I to respond to his anger? Was I supposed to show him the design equations for my piezo tweeter, the dispersion polar plot, or the frequency response of the monitor as measured in an anechoic chamber? No, I'll not find answers to my problems in the engineering literature.

• **Nov. 7, 1983.** Where contemplation has failed, maybe facts will succeed. I will try to jot down true things about monitors: Flush-mounted monitors give a theoretical 3 dB increase in output as compared to mounting in free air. (They also look more professional.) If a monitor is mounted with the bass unit at the bottom and the high frequency driver at the top, a tall person will have a shorter path length to the high frequency driver whereas a short person will have a shorter path length to the woofer...unless they are both sitting down. It is not unusual to move one's head a few inches from the operating position and hear a totally different response characteristic; the console's reflection also induces a comb response. Thus the best listening position is with the head nailed to the console armrest. The best way to evaluate a monitor is to listen to previously recorded tapes with which the engineer is familiar. You know what's on the tape because—well, because you've heard it so many times that you just know. Or, the best way to evaluate a monitor is to compare it to live material. You *do* know what the inside of a kick drum sounds like, don't you? The really best way to evaluate a monitor is under anechoic conditions, using test tones. Of course, you can only use them for steady-state mixdowns in anechoic chambers—a wall or floor would probably make sound boomy and lacking in resolution. Actually, you should sweep the monitors with pink noise (plug your ears, it should be loud) and watch an RTA, diddling with equalizers until it looks okay. No, really only a TEF test will do the job. Make sure the absolute phase switch on your power amplifier is switched to either positive or negative. Well, the general idea is to get the most clinical response at a spot two feet behind the mixer's head and three feet to his left—where the producer is sitting. This guarantees that you have fooled the producer and he will recommend that the record be shipped triple platinum (make sure you have points on the deal).

• **Nov. 8, 1983.** No, perhaps even facts fail to address this problem. Sure, I can measure specifications, see how much level they can take, even get a feel for listener fatigue, but as far as an evaluation of the sound—apparently everyone has their own idea about what a monitor should sound like. The point is that monitors are not consistent, and not remotely close. While an untrained ear might not be able to discern between vacuum tube and solid state amplifiers, almost everyone can hear the difference between almost any pair of monitors. The differences are too great to obscure in equations, differing testing procedures, different rooms, or anything else. I am convinced that the problem must be approached from a subjective standpoint, in terms of listener context and listener preference.

• **Nov. 21, 1983.** The problem is, it seems impossible to say anything tangible about a topic as intangible as a listener's evaluation of a monitor; it's probably more a question of fads and one-upmanship than anything else. I really don't know what I'm going to do as far as this test report goes. On the positive side, the monitors the magazine wanted me to evaluate haven't showed up yet. Perhaps the factory was hit by a big meteor or something—one never knows. I am keeping my fingers crossed.

• **Nov. 28, 1983.** I visited a "world class" studio this morning, looking for answers. They showed me their control room—truly an artistic triumph. They assured me that the acoustics were perfect, "as good as money (and a lot of it) can buy," the studio manager joked. They were anxious to have me audition their "professional standard" monitors; they put on a tape and played it for me. I listened carefully, and wondered what was wrong with my ears. The sound seemed to have an unpleasant honk to it—like a foghorn or something. I asked someone about that, and he assured me that I simply wasn't used to true response. He pointed to a rack of 1/3 octave equalizers: "You see, everything has been carefully adjusted with an RTA." I wandered forward and peeked under a grille cloth—there were three pairs of flushmounted monitors over the window. I asked them which was the perfect pair to match the perfect room. They didn't think that was funny. I asked why they needed three pairs, and how many consumers owned such custom-designed systems, and how many consumers listened in a million dollar room and why the producer, who had just entered, was bringing his own Radio Shack models with him? They hustled me out and told me not to come back.

• **Dec. 5, 1983.** The speakers still haven't arrived. It is too soon to celebrate, but perhaps this whole affair will simply blow over.

• **Dec. 9, 1983. (A.M.).** I am in a hi-hi-price, hi-fi store, listening to their monitors, or, as they call them, loudspeakers. I am very impressed; they sound a lot better than the ones in the world class studio. But how could a consumer loudspeaker sound better than a professional one? I've heard rumors about this sort of thing—that some of the small "golden ears" audio companies are making equipment that is better than their professional counterparts. Some people say that for everything from JFET preamplifiers to electrostatic loudspeakers, the consumer companies have out-maneuvered the more traditional professional manufacturers. One thing is clear as I sit in their comfortably-appointed living room-type listening room (what a strange idea—to audition music and equipment in a room similar to the ones in which people actually listen to music), the fidelity is very good. This pressing of a symphony orchestra sounds much better than any classical sound I've heard in a commercial control room. Ah, perhaps that is an unfair comparison, since control rooms are designed to play back pop music....

• **Dec. 9, 1983. (P.M.).** On the way home from the hi-fi store, I suddenly thought of another problem. The SPL levels were much lower in the store, as compared to most mixdown rooms. This raises other questions—the entire psychoacoustic thing—such as the Fletcher-Munson equal loudness contours. Since our hearing response changes so radically with respect to phonic level, how can engineers even begin to design monitors? To make any sense of it, they would have to rate monitors (this monitor system has been calibrated for flat response at 115 dB SPL at the mix center?). Or should we attempt some sort of microprocessor feedback loop in which a sense microphone monitors the room SPL levels and adjusts outboard equalizers to compensate for human hearing change with respect to loudness? Or why bother? I guess the engineer's hearing was also human....The more I get into this, the more hopeless it becomes.

• **Dec. 10, 1983.** The monitors are here. When I went into the studio, there they were—crouched in front of the console. Someone had set them up, as if to personally belittle my plight. But I was too clever for them. I told them I had a head cold—couldn't hear a damn thing—and ran home and locked the door.

• **Dec. 15, 1983. (A.M.).** I can't put it off any longer. Today is the day. The magazine called and said I either had to listen to the monitors or send back the magazine's advance of $10—long ago spent on aspirin.

• **Dec. 15, 1983. (P.M.).** I am in the studio now, face to face with the task. The speakers confront me—eighty pounds of impossibility. There is a tweeter and a 12-inch woofer (crossover at 1,700 Hz) installed in a vented enclosure with Butterworth tuning. I have placed the monitors against the wall, five feet off the floor on a wooden shelf; the three position tweeter attenuator is set to flat. I have instructed my second engineer, Jose Vazquez, to load a master tape so we can listen to clean tracks. What good would a mixed tape be for evaluation—its construction already irrevocably poisoned by some other monitors?

We begin with a bossa nova tune. Instantly the high efficiency of the monitors is apparent. We proceed, track by track. Cabassa: up front, too much sizzle, hard, not pleasant, too hard—metallic. Triangle: strangely lacking in ambience, too metallic again. Low quiro: a little edgy but okay—apparently we have come down from an upper frequency problem area. High quiro: the edge returns—not pleasant. Electric guitar: not too bad of a sound—upper mids have an aggressive edge again, but it is not as unpleasant on this instrument. Bass: no low bottom at all—the bottom octave is gone—as if the woofer isn't moving any air. The sound is more an upper, tighter sound. Jose likens it to the output jack on the Rickenbackers that adds higher frequencies. Flute: tons of presence; the mid range peak of the monitors is overwhelmingly apparent now—yet the flute sound is good because of it—an incredibly live sound. Strings: the peak works against us here—tinny, middy—sounds like glass violins with glass strings and steel horsehairs.

Jose Vazqeuz changes tapes—we try a disco tune next. Brass tracks with three bones: tinny, not round, not mellow. Piano: an interesting sound that would work in this mix; very percussive, a lot of hammer sound—that 6 to 8 thousand Hertz boost again. Kick drum: lots of punch; these are very punchy monitors—a good tight sound—but the lack of low bottom is apparent. Hi hat: a great sound; that sizzle makes it happen—even at moderate listening levels, you involuntarily blink your eyelids in time with the hi hat. Cymbals: likewise tremendous presence; I can feel my hair being parted along the top of my head.

Jose changes tapes again and we get serious this time with a hot techno-wave tune. Everything is synthesized, or sounds like it ought to be. Synthesized drums: the lost bottom isn't as big of a problem—the incredible punch makes up for it. Hi hat sound: mucho sizzle again—punchy. Synthesized guitar: this is interesting; a strange nasal sound has been added. Perhaps the overall defect of the monitors is at last revealed—the sound you make when you talk and cup your hands over your mouth. On this guitar it is strangely pleasant—a vocal sound emerges. However, Jose disagrees; he liked the original track better. To

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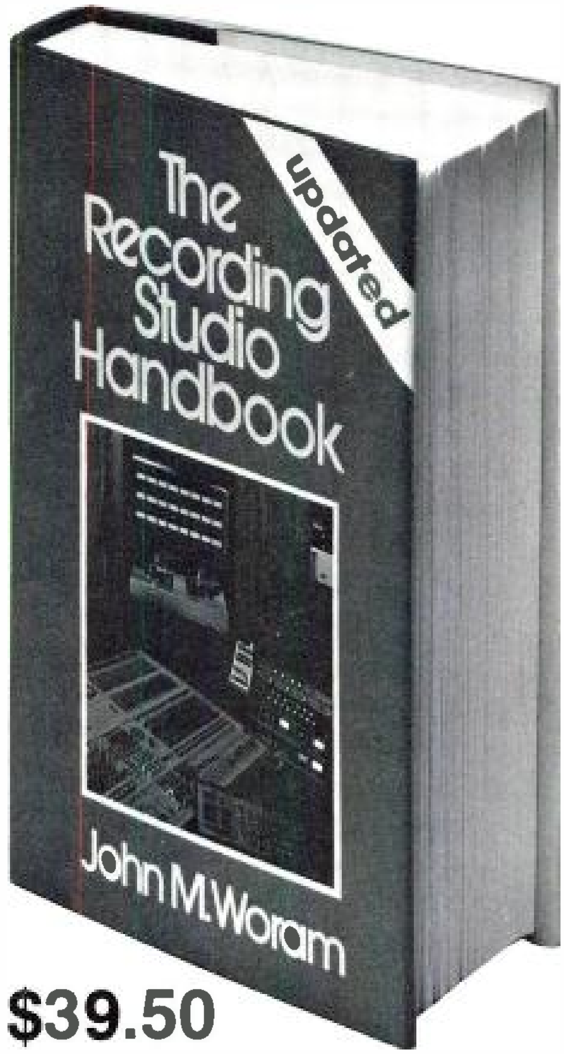

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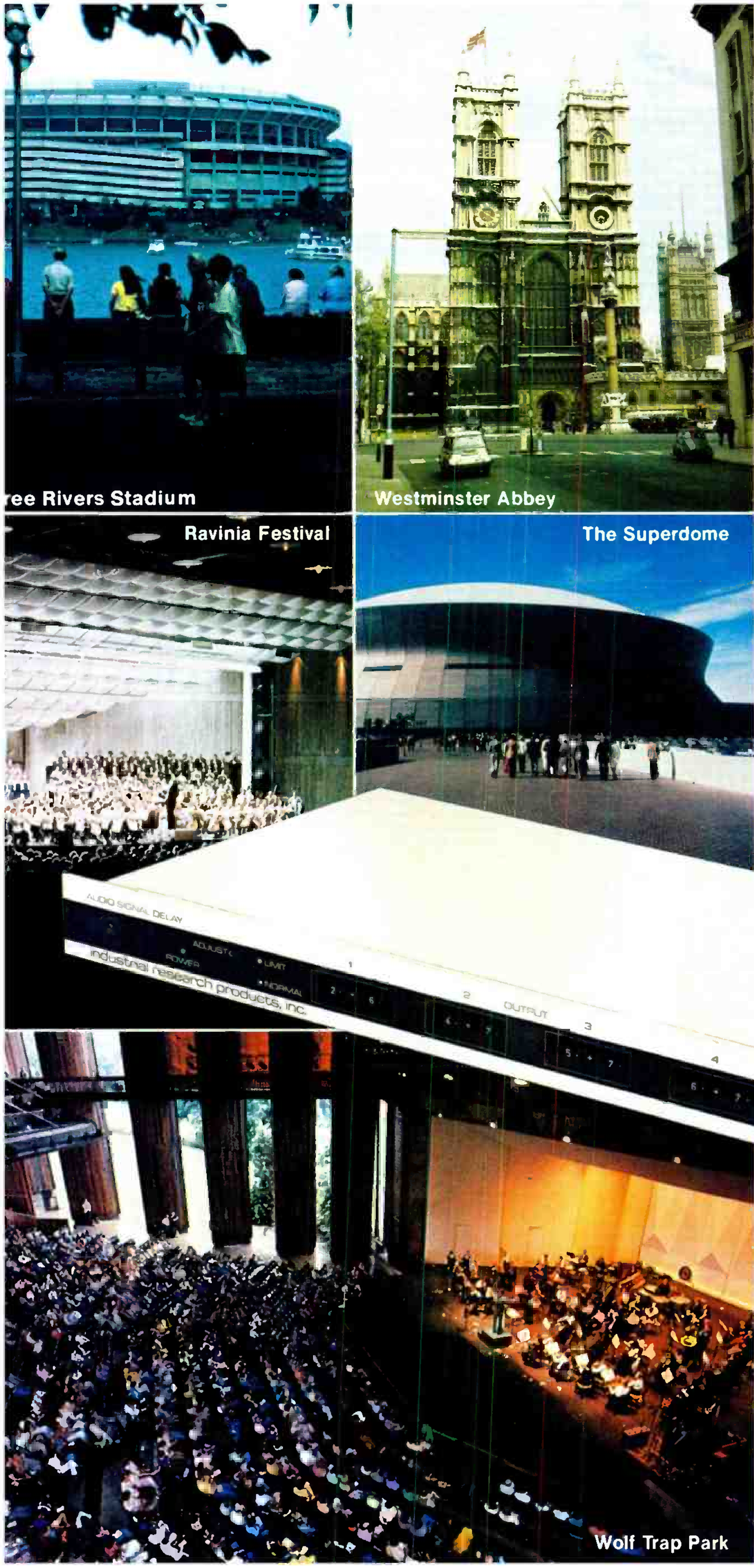
hell with him—I'm the first engineer around here. Bass: lost bottom again. In a pop tune, where the bottom probably wouldn't have been mixable anyway, this sound might have been great—its cutting edge up-front sound would make it mix through anything. But for this techno-wave tune, the mix would lack support. Vocals: this is the toughest question on these monitors. Finally Jose and I decide that they are unnaturally up front, as if the monitors supply an automatic vocal mix. It's incredible to hear the way the vocals come out from the mix—unfortunately, that is not the way the tracks really are.

We stop tape, after two and a half hours of loud listening. Our ears are going, and the sizzle from the monitors sounds milder to us now. Like all good engineers, we compensate like crazy and keep listening—we experiment with the high frequency attenuation control. A -3 dB setting helps the hot high frequency problems, but has no effect on the upper mid peak. We move the monitors into the corners, and listen to the bass—it has been augmented, but the low bottom still isn't enough. We mess around—toilet paper rolls stuffed in the ports, place them on the floor, play with the high frequency attenuator, and find that the monitors can be worked with easily and some of the problems corrected. However, the low bass cannot be resurrected, and the nasal middy AM radio-type boost remains. Of course, the process of roughing-in a monitor is long and involved, but we conclude with the distinct evaluation of a very up-front speaker—a monitor with real bite, and overbite too. But my old doubts and fears have sneaked up on me again—the idea that this monitor, and all the rest of them, do more to the signal than all the signal processing gear in my rack.... My head has started to ache again; I start getting shaky. When Jose isn't looking, I escape.

Okay, I listened to the damn things. Actually, they weren't too bad.... It's probably just as well the meteor missed the factory. However, I didn't like them as much as my old monitors—no, what am I thinking? How can anyone mentally compare monitors? It's crazy. Even direct A/B comparison depends on the room, program material...and here I am, introducing imagination and memory into the comparison. And what about my report? What am I supposed to say to the readers? Write down guitar, and say—too nasal? My subjective opinions are good only for me. Anyone else could hear them totally differently. Should I run free field response curves, measure directivity, and send that to the magazine? That would be equally useless. Must each individual merely listen, and make up his own mind, siding with the current consensus, or what? Oh, I don't know, my headache is coming back....

• **Dec. 19, 1983.** No, I've made up my mind. They won't get anything out of me. What I think about a monitor is my own business. Ah! What's that kick drum sound? It's the magazine people, pounding at my door—trying to get in—trying to get my monitor evaluation! Ha! Ha! They're too late—I'm cramming these pages into my mouth, chewing frantically—they'll never get it.... ■



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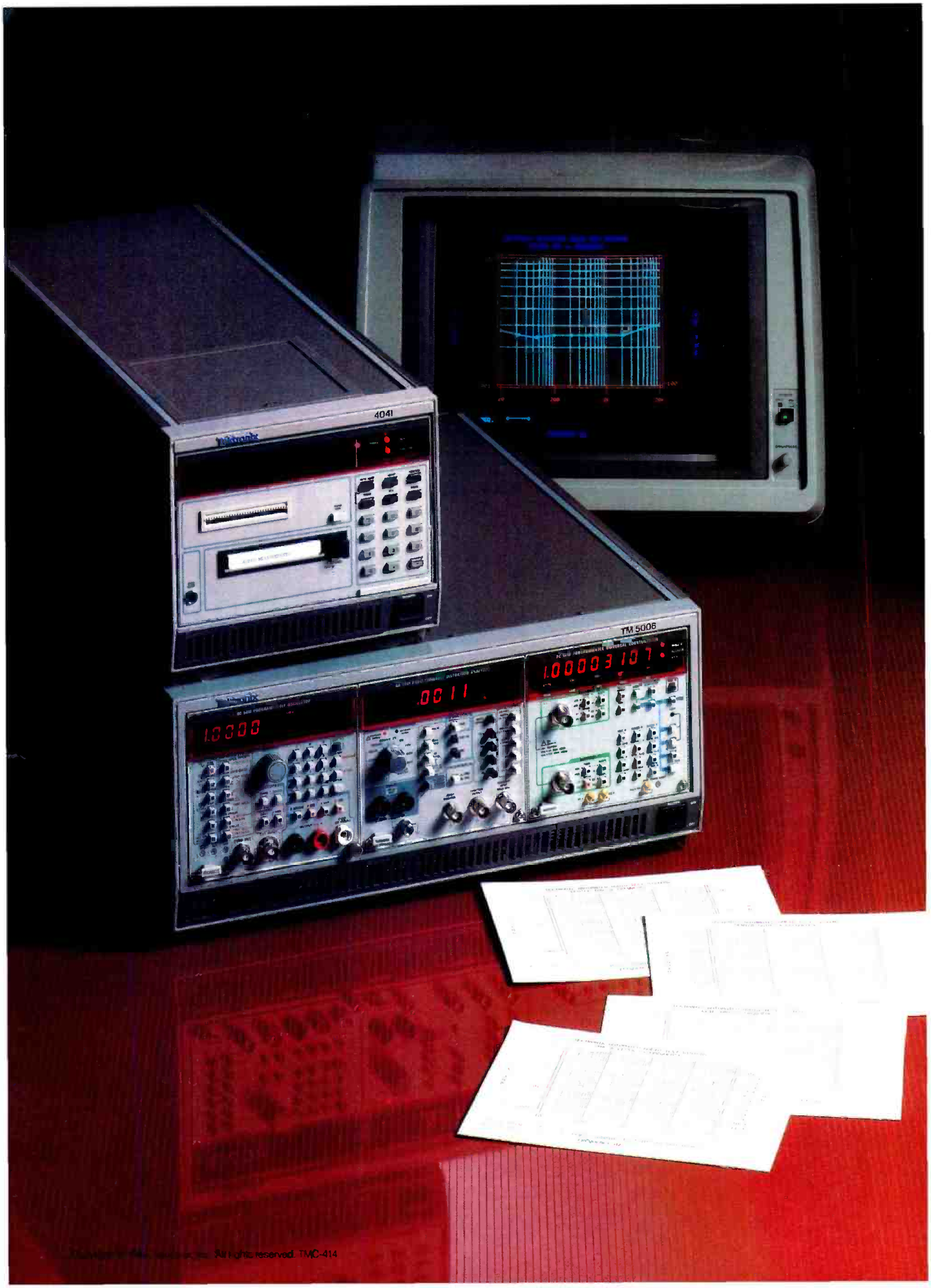
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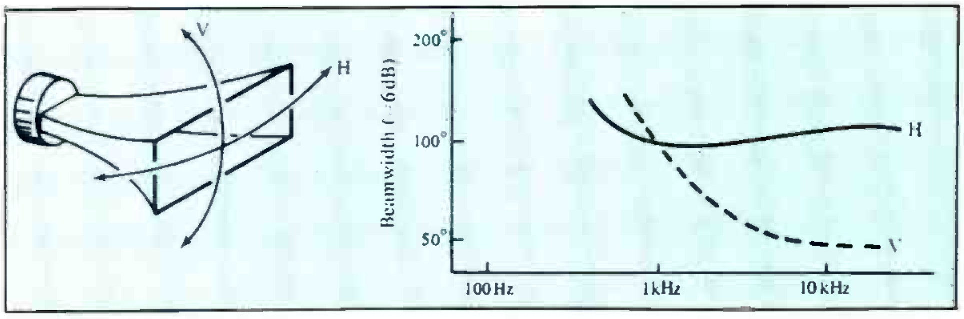

Figure 1. Beamwidth plots for a 90 x 40 degree horn.

In-line stacked arrays can be used to provide narrower coverage in the vertical plane. In this application, the large effective vertical mouth dimension provides tighter pattern control in the vertical plane at lower frequencies.

TECHNIQUES FOR WIDENING COVERAGE

FIGURE 1 shows the vertical and horizontal beamwidth of a single horn. In this case, the horn has nominal 90 degree horizontal coverage and 40 degree vertical coverage. Beamwidth is defined as the angular spread between the -6 dB points, relative to the on-axis value. When a pair of these horns is stacked and splayed 90 degrees, as shown in FIGURE 2, the effective horizontal angle is extended to 180 degrees. Note that the splaying has taken place along the nominal -6 dB zones of the two horns. If they had been splayed at an angle less than 90 degrees, the coverage would have exhibited a "bulge" along their common axis. If they had been splayed at an angle greater than 90 degrees, there would have been a dip in their coverage along their common axis.

By extension, a horn with a nominal 90 degree horizontal coverage angle and a horn with a nominal 60 degree angle can be splayed along their -6 dB zones to create an effective horizontal coverage angle of 120 degrees. Obviously, many other combinations can be used similarly.

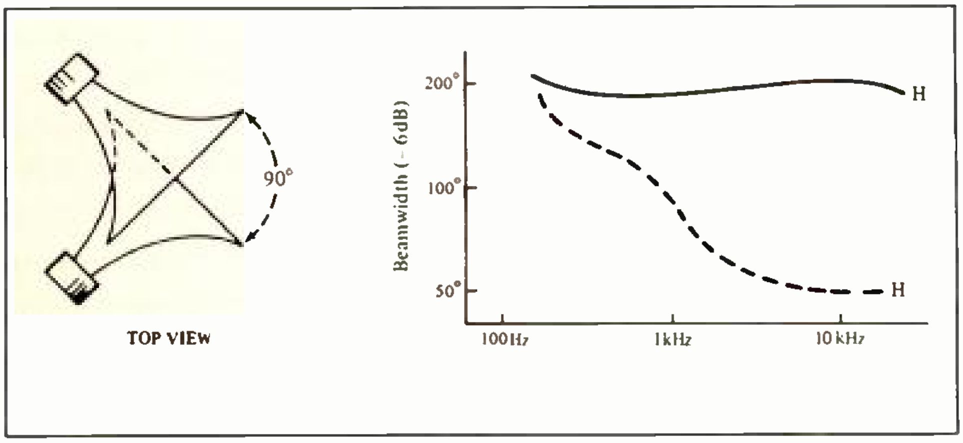

Figure 2. Two 90 x 40 degree horns stacked and splayed 90 degrees.

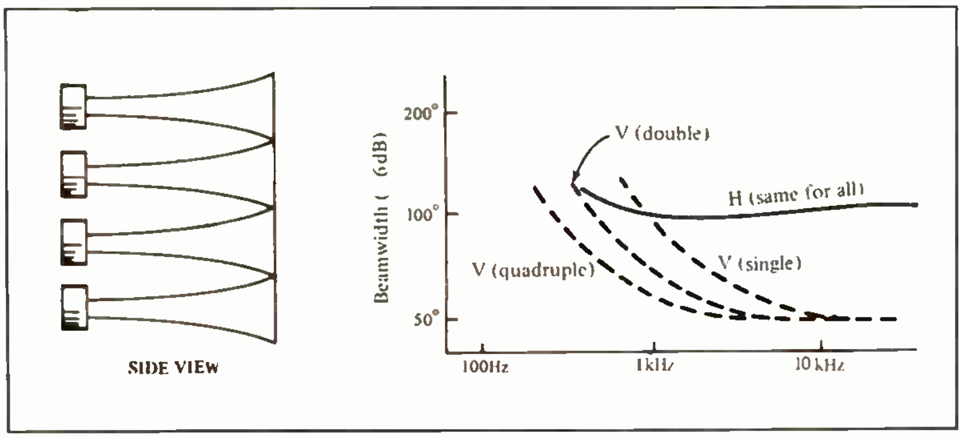

Figure 3. Vertical in-line stacking of horns.

Note that the vertical coverage of the stacked array is substantially the same as with the single horn.

It is important that the horns be stacked rather than placed side by side. Stacking minimizes the effective combined mouth dimensions, resulting in less interference between the horns. If the horns were placed side by side, the effective horizontal mouth dimension would be twice that of a single horn, and there would be some irregularity in response—lobing—in the horizontal plane.

TECHNIQUES FOR NARROWING VERTICAL COVERAGE

As we saw in FIGURE 1, a typical 90-by-40 degree horn maintains its horizontal pattern control to below 1 kHz. The reason for this is that the horizontal mouth dimension is fairly large, as is the coverage angle. In the vertical plane, however, pattern control is lost below 2 or 3 kHz. The reason for this is that the vertical mouth dimension is small, and the desired coverage angle, 40 degrees, is also small. If the horn were designed with a vertical mouth dimension, say, twice what it is, we could then expect the vertical pattern control to be maintained an octave lower. Through vertical in-line stacking of horns we are in effect creating a larger mouth dimension, and we can maintain pattern control to lower frequencies. This is shown in FIGURE 3.

When we do this, the horizontal pattern of the vertical array is exactly the same as for a single horn. Vertical control is maintained to lower frequencies, halving for each doubling of horns in the stack. Usually, three or four horns represent a practical limit, since there will be some degree of interference between the sound sources. Lobing is normally less than 6 dB, and we can usually ignore it.

Such stacks are useful for long-throw applications in speech reinforcement in highly reverberant spaces. The multiple vertical stack will usually result in an array that has little depth to it, compared with a single low-throw 40-by-20 degree horn that may be in excess of four feet in depth. ■

BARRY BLESSER

Phase Locked Loops Continued

• Just like the afternoon soap operas, you, dear reader, will not be able to follow the story line if you did not read last month's column. So go find your copy or "borrow" your friend's. We introduced the concept of the phase parameter and showed that a periodic signal, like a sinewave, could be thought of as a ramp in phase. We demonstrated a circuit technique for taking two input signals and producing an output which was proportional to the phase difference of the inputs. This element, the phase detector, is one of the two critical blocks in the phase locked loops. The other one is the voltage controlled oscillator.

VOLTAGE CONTROLLED OSCILLATOR

This circuit element can be built in many ways, and it exists just as a monolithic IC all by itself. Almost any oscillator or function generator can be turned into a VCO (voltage controlled oscillator) by varying one of the parameters in the circuit (e.g., current, voltage, resistance, etc.) A formal mathematical description of such an element might be in any of the following formats:

(1) $\text{Freq} = k \times v$

(2) $y(t) = \sin([kv]t+\theta)$

(3) $\theta = k\int dt;\ y = \sin(\theta)$.

Of these we will use only the first part of equation (3). This definition is the best for our purposes because the variables in the description are the input voltage to the VCO and the output phase. Remember, we are considering the phase variable in the system, not the actual signal. Equation (3) is very simple when expressed in words. It says that for any constant input voltage, the phase will be a ramp whose slope is proportional to that input voltage. The scale factor k is an arbitrary constant determined by the particular VCO being used. Why have we made this trivial issue appear to be important? The answer is that equation (3) is a very primitive filter. The integral means that, in circuit terms, the two variables can be simulated by an integrator. The circuit model of FIGURE 1 has the identical characteristics as that of a VCO using the phase variable! We can thus say that a VCO is really a simple op-amp integrator.

We are now in the position to show you a phase locked loop.

PHASE LOCKED LOOP

The two parts of FIGURE 2 give a system block diagram of a phase locked loop and an equivalent model of it in linear terms. Notice that the model below is a very simple linear filter circuit with feedback. If you have the mathematics to understand how to find the system function of the model, you can also predict the behavior of the actual phase locked loop. The system function can thus be shown as follows:

$$(4)\quad \text{Gain} = \frac{\theta_2}{\theta_1} = \frac{F(s)}{F(s)+s}$$

where F(s) is the system function of the lowpass filter. For those of you who have no idea what all this means, simply pretend that it is some nonsense. Those who do understand the system function idea should be impressed by the fact that a very complex nonlinear time-varying feedback loop can be analyzed as a simple filter system.

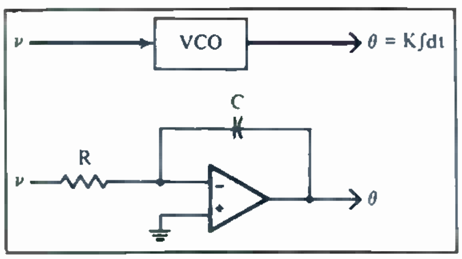

Figure 1. A VCO (top) and a circuit model.

In any case, it is now time to give both classes of reader a physical feel for the behavior of the loop. We begin by considering that the VCO is operating at the correct frequency as the input signal, with the correct phase difference between the two signals. The phase detector puts out some DC value which goes through the lowpass filter unchanged and unamplified. This DC keeps the VCO at the correct frequency. Now consider what happens if the input frequency is increased just slightly. After one cycle, the two inputs to the phase detector have begun to drift apart. As they slide relative to each other, the phase output signal begins to increase. This information goes through the lowpass filter and appears at the input to the VCO. The VCO now increases its output frequency in an attempt to catch up to the input.

In other words, the output frequency, and phase, will track the input frequency and phase. At this point you should notice that we have taken a great deal of trouble to build a circuit that really does nothing. Just be patient. The circuit actually does something very interesting—but not for us. It is an FM demodulator! Instead of taking the output from the VCO output, we could take the output from the lowpass filter. This signal is a voltage that creates a frequency equal to that of the input. When the input frequency goes up, this voltage goes up; when the input frequency goes down, the voltage goes down. If the input were an FM modulated carrier, then the output would be audio. Unfortunately, we are not interested in FM demodulation. We started this project because we had the problem of clock generation.

Now back to the issue at hand. We will now make a small change to the VCO as shown in FIGURE 3. We have broken the VCO into two sections: a higher frequency VCO followed by a digital divider (counter). As a black box, the new VCO looks like the old one. Assume, for example, that the old VCO produced 50 kHz output when the input was 3 volts. We select the new VCO such that 3 volts produces 500 kHz and the digital divider has a factor of N = 10 (divide by 10). The output is also 50 kHz. The phase locked loop cannot tell the difference. Since the output will track the input, the 50 kHz from the

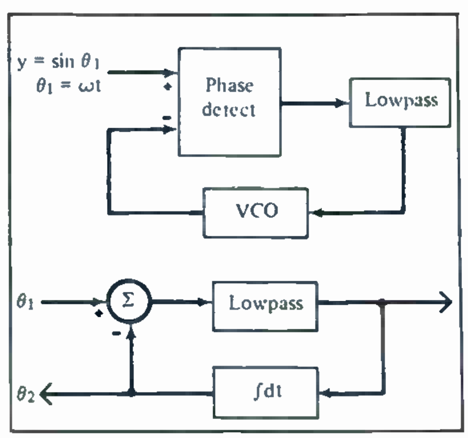

Figure 2. A phase locked loop (top) and a linear model.

VCO will exactly match the input 50 kHz. However, when this match happens, the internal frequency of the VCO will be exactly 10 times the output. We have made a frequency that is a factor above the input. This is a frequency multiplier.

Presto! We can now take the input clock of 50 kHz and create a local oscillator frequency of 500 kHz. The two frequencies will always be synchronous and they will be phase matched. We have thus solved the problem. In the local equipment, the 500 kHz allows us to have unit intervals of 2μs. Exactly 10 of these units will fall in the period of the 50 kHz input.

If the input frequency were 50.0001 kHz instead of the exact 50 kHz, the internal clock would become 500.001 kHz. Every frequency and time interval would stretch to the same factor.

TYPICAL APPLICATION

A typical application for this technique might be the following. A digital studio has many different types of digital equipment. All of the equipment is run from a common clock, called the master. The A/D converter module on one of the microphone inputs needs to produce a digital word on every clock cycle of the 50 kHz reference master. However, internal to the A/D we find a sample-hold amplifier and a successive approximation register. Assume that the sample-hold amplifier requires 4μs for acquiring the signal and that the successive approximation algorithm requires 16 steps. The 4μs for the sample-hold leaves us 16μs for the 16 steps. This would suggest that we need a 1 MHz local oscillator to produce 1μs intervals. The sample hold would run for four such intervals and the 16 steps of the conversion would each have 1μs. We thus design the VCO to run at 1 MHz and use a division by 20.

The problem gets slightly more complicated if the ratios are not simple. Consider an example where we wish to convert a 44.1 kHz clock into a 50 kHz clock using the phase locked loop. This problem was mentioned several months ago (October '83) when we discussed the issue of sampling rate conversion. We could use the same technique if there were a digital divider that could divide by 1.1337868. However, there is no such thing. But all is not lost. Notice that the relationship between the two clocks can be expressed as 500/441. Suppose we place another digital divider on the 44.1 kHz input and set the divider factor to 441. The result will be a frequency of 100 Hz. Now suppose we set the VCO's divider to a factor of 500. In order to produce the same 100 Hz, it will have to run at 50 kHz. This illustrates that the phase locked loop can multiply by any ratio of rational factors. The phase detector system runs at a frequency that is the lowest common factor of the input and output frequencies. When used in the sampling rate converter, the divider ratios become the word count ratios. For every 441 words which enter at 44.1 kHz there will be 500 words which exit at 50 kHz. We also see that by reversing the two divider numbers we could just as easily convert in the other direction.

Back to your mental garbage can. It is time to file the idea that the phase locked loop can be used as the basic element in a frequency synthesizer. By programming the count numbers we can create fixed relationships between different frequencies without any complexity. Many frequency synthesizers are actually built this way.

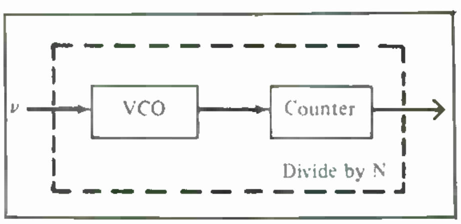

Figure 3. A new VCO.

ISSUES

Before you run off to build such a system, keep in mind that there are many important topics that we have not considered. The phase locked loop is actually a very complex system when one wishes to consider other variables. Many students have earned their Ph.D.s just by studying some very small aspect of the phase locked loop. Some of the topics we have ignored are the following.

When the system starts, the simple model of FIGURE 2 does not predict the dynamics of coming into lock; the model only predicts the behavior once lock has been achieved. All of the model elements break down and the locking behavior is very nonlinear. In some special applications, like satellite tracking, the loop might not actually ever lock. It requires special capture circuits.

The dynamic range of the phase detector is limited. It is therefore important to predict when input signals will result in overloading the limited range.

For digital audio, the most important side effect is that the output frequency will have some jitter even if the input is spectrally pure. The output has phase noise. This means that each clock will have a slightly random deviation even though the average deviation will be zero. The loops keeps the averages correct but not the instantaneous values. For example, with a 1 MHz VCO locked to 50 kHz, the clock transitions might be at the following times:

0
1.001μs
1.999μs
3.000μs
4.011μs
5.004μs

This kind of jitter might result in audio noise if it was used to control an A/D or D/A converter. In a much earlier article we indicated that the clock had to have jitter less than 1 ns in order to prevent degradation at 16 bits with a 50 kHz clock. The above table has jitter of 11 ns peak. As it turns out, it is extremely difficult, if not impossible, to build such good quality VCOs with such extremely low jitter. The best possibilities appear to be with VCOs built with crystals, but these have a very limited frequency range. In simple terms, it is best not to have a phase locked loop in this one critical area. The A/D and D/A should be run directly from the master clock with no other processing. The phase locked loop can be used, however, in purely digital equipment. A digital reverberation system might need an internal 20 MHz clock in order to create 400 program steps for each input sample. That would be a perfect application for the phase locked loop, since it could create a frequency multiplication of 400. Small phase jitter would play no role, since each word and program step is based on a counting—not a linear time. As long as the computed result appeared 400 steps after the input, it would not matter if the 400's steps were off by 50ns as long as the average rate was exactly correct. ■

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ARE WE MONTHLY or bi-monthly? A good question since we had recently sent you a Nov./Dec. issue, followed by a Jan./Feb. issue, and here we are with a March issue.

Well, we're still a monthly, but one that, frankly, got behind on our issues. Now we are almost completely caught up.

What will this mean if you are a paid subscriber? Will you get the twelve issues you are entitled to? The answer is absolutely. The computer is programmed for the number of issues and will override the expiration date if it gets to that first.

There will be monthly issues from here on out. This month, the subject is Sound Reinforcement, and just a quick look at the table of contents will convince you that it is just that.

We're doing exciting things with **db**. You'll notice more articles—and more color. If you look at the cover of this issue and the layout of the articles you'll see some changes in graphics. But you'll also notice that the articles are still the same incisive and detailed discussions you've come to expect in our pages.

Watch us grow!

* * *

Digital—It's here to stay. As this is written, I'm off on a trip to Japan to Matsushita to see all their new digital and analogue toys. From Japan, in a rare burst of masochism, I fly directly to Miami, Florida, for the SPARS Conference on Digital Electronics. As they say, watch this space for further developments! L.Z.

MARTHA L. FISCHER

Dallas Sound Lab: The Inside Story

Dallas Sound Lab, a 48-track facility serving the film, video, and recording industries, is an integral part of the multi-million-dollar Dallas Communications Complex.

WHEN MOST 24-CHANNEL audio studio owners look to expand their business, they think in terms of adding more equipment, renovating existing space, or at most moving across town. Russell Whitaker, however, thought in terms of selling his studio in Austin, Texas, moving his family some 200 miles north to Dallas, and building a brand new facility from scratch. Why?

"Opportunity," said Whitaker. "A chance to enter the Dallas market at a time when the entertainment industry was building momentum there. And especially, a chance to become an integral part of the Dallas Communications Complex."

The Complex, a four-building communications center that started construction two years ago in the Dallas suburb of Irving, has received considerable attention. Finance and startup of the development by commercial real estate magnate Trammell S. Crow was based on the concept that a sophisticated multibusiness facility serving a wide range of communications needs would attract both quality tenants and celebrity customers. To date, the former list is growing, while the latter includes: ABC Motion Pictures (*Silkwood*), Porthos Productions (Robert Altman's *Streamers*), NBC Productions (*Celebrity* mini-series), Stevie Wonder, Kiss, David Bowie, Robert Plant, Stevie Nicks, Genesis, Eric Clapton, Jimmy Page, Jeff Beck, Jan Hammer and Joe Cocker.

As the only audio facility in the development, Dallas Sound Lab looks to capitalize on the prestigious location. Design, of course, follows function, so Whitaker decided to focus initially on his service objectives.

1. *Live audio recording from the sound stages.*
   Film and video projects frequently require multitrack audio for recording live concerts, orchestra music, or multiple sound effects during a shooting session in the sound stages. An extensive interconnect system would be needed to hardwire the miles of cable required from the three sound stages in Building One to Dallas Sound Lab's control room in Building Four, some 250 feet away.
2. *Audio post-production for film and video.*
   Facilities would be needed for assembling sound effects and music to film or video picture, along with voiceover, dialogue replacement and sweetening capabilities. Film dailies would need to be viewed in a theater equipped with a large screen, ample room and comfortable seating. A 4-channel Dolby sound system would also be needed to screen finished features.
3. *Jingle and commercial production services.*
   To accommodate large-scale commercial production, the studio had to be big enough to hold strings, brass, woodwinds, percussion, drums, bass, guitars and

Martha L. Fischer is a free-lance writer working out of Dallas.

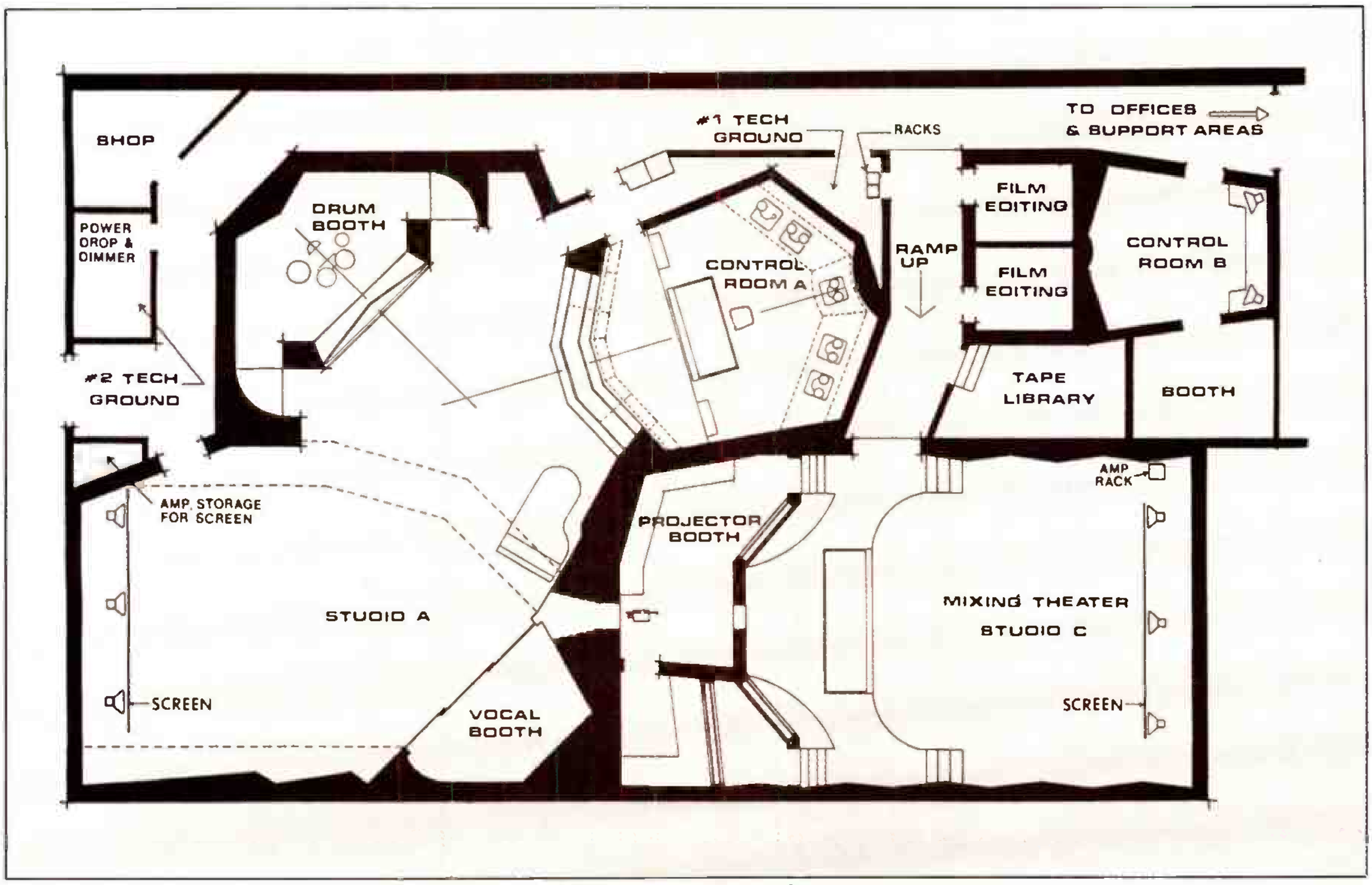


Figure 1. Floor plan of Dallas Sound Lab.

Figure 2. The view from Control Room A, looking into the studio.

vocals simultaneously. Also, the control room design had to be ample enough to seat a sizable contingent of agency, producer and client personnel.

4. *Dual film projection.*
   While a commercial was being produced in the main studio, rushes for drama might be screened in the film mixing theater. A dual-sided projection booth would be developed to service both rooms independently. The studio recording screen would have to be positioned in such a way that the musicians could see it as well as the people in the control room.

5. *Music recording for singles albums, videos and demos.*
   Groups generally prefer to record close to the control room, to maximize rapport with one another and to see visual cues from the producer. Live music video shooting capability would need to be established with pre-wired camera lines, an extensive lighting and grid system, ample studio room, and access to the film screen for front projection effects. The drum booth needs to be in close proximity to the control room, and the recording area directly in front of the window should be relatively "dead."

Having a low noise floor is especially critical these days, with the increased popularity of distance mic'ing. As more ambient room sound gets on tape, it is important that the room be as quiet as possible and free from outside intrusions.

In a session, the lights get hot and the cigarette smoke tends to build up. Musicians demand comfort, and the HVAC (heating/ventilating/air conditioning) systems must operate almost continually. It is unacceptable to simply shut off the air conditioning for certain music passages, as had been done in earlier times. So the design elements must be increasingly sophisticated to take these factors into account.

From the design point of view, a goal of extremely low noise criteria was established (NC 20) with mechanical and HVAC systems operating. In addition, the overall design needed to be visually pleasing and functionally workable in terms of traffic and crowd control. For example, while one group moves equipment out the front door, another moves smoothly in through the back door.

Even color coordination was determined well in advance. Whitaker decided to move away from the traditional earth tones into the more contemporary "high tech" colors of blue and grey with silver accents. The exposed lighting, designed by Tully Weis of Dallas, is operable from the studio and control room by an extensive solid state dimmer system.

Beyond lighting and color, atmosphere is determined largely by the sense of privacy that each client receives. This principle holds true for virtually any establishment, from a high class restaurant to a high quality studio. Each client must enjoy a sense of isolation from other activities in the building—especially in the recording industry.

DESIGN CONSIDERATIONS

Whitaker put his initial thoughts on paper in January, 1982. Six months later, he brought in Johnny Marshall, a former co-producer and keyboard player with extensive studio management experience. Marshall's knowledge of the Dallas music market was a key asset to Whitaker. They spent another five months together refining their ideas.

After reviewing preliminary designs from a number of sources, Whitaker contracted with the acoustical consulting firm Joiner-Pelton-Rose, Inc. in December, 1982 for studio design. Construction began in January 1983, and the official grand opening occurred this past November 18th.

Senior consultant Russell E. Berger II supervised the project and was responsible for overall acoustical and functional design. Berger acknowledged several associates for their involvement: Jack Wrightson for psychoacoustics analysis; Tom Rose for transmission loss of partitions and HVAC analysis; Topper Sowden for grounding systems and electrical power noise control.

The first challenge emerged immediately—the "conveniently" close proximity of the Dallas-Fort Worth airport. On-site testing revealed that the peak levels measured a full 96 dBA. So to achieve the desired NC 20 noise level within, the exterior of the building had to provide a 75 dB transmission loss from outside to inside. A further complicating factor was that most wall sections and ceiling partitions are rated according to the standard Sound Transmission Class (or STC) rating, which is inappropriate except as a rough guideline.

The rating system, originally adopted and designed to control speech-range noise problems, is woefully inadequate for the sensitive low frequency criteria a studio requires. "While the STC rating may be fine for a contractor concerned with how much human speech may leak through from one interior office to the next," Berger said, "it does not provide information on the wider spectrum of noise a studio designer must take into consideration."

Fortunately for Whitaker, Joiner-Pelton-Rose had completed a major acoustics study for the city of Dallas concerning projected DFW noise levels for the next 50 years. The data base proved very useful for this project.

Next, the firm studied the possible impact of present and future neighboring tenants in the building. Dallas Sound Lab takes up some 8,000 square feet of an existing 500,000 square feet. HVAC units located on the roof would also add vibrational and structural noise. Once all these potential noise sources were identified, the firm again reviewed STC curves and the spectra of the wall sections and partitions under consideration. Selection of materials was critical.

The solution was based on a single premise: mass stops low frequency energy. So high-mass materials such as masonry and concrete were utilized to a considerable extent. Drywall partitions incorporated large dead-air spaces. Wherever possible, walls were decoupled; where bridges or connections already existed, they were resiliently isolated.

Floating floor systems were implemented for both the control room and studio areas. The roof system is floated on resilient spring and rubber isolators, and is completely independent of the side walls. Interior side walls are built upon the floating floor system. This effectively decouples the interior from the exterior of the building and outside sounds.

Rooftop vibrations from mechanical HVAC systems located there created a separate problem. Vibrations travel from a roof down the support posts to the ground—which the slab rests upon and the walls are built around. These noises, both vibrational and airborne, required widespread decoupling as a means of sharply limiting their effect.

In fact, the mechanical units themselves had to be addressed directly. Each rooftop unit, both on Dallas Sound Lab and on neighboring tenants, is floating on resilient spring mounts. The wall sections adjoining neighboring tenants were also designed to handle 96 dBA, and were thus quite massive. Partitions incorporated various combinations of masonry/masonry and masonry/drywall.

Structural noise was by no means the final obstacle to be overcome. Berger said an "antenna farm," with satellite uplink and downlink capabilities, would be located nearby. This presents, of course, a serious radio frequency (RF) hazard to low level audio signals.

INSTRUMENTS

Steinway 9′ concert grand
Hammond B-3 with Leslie
Yamaha DX-7 digital synthesizer
Yamaha C5-50 polyphonic synthesizer
Prophet 5 polyphonic synthesizer
Prophet 1005 poly sequencer
Mini-moog model D synthesizer
Hohner Clavinet
LinnDrum
Roland Bass line
Roland Drumatix
Ludwig Drum Kit

EQUIPMENT

Audio Tape Machines:
Otari MTR90 .... 24 track
Otari MTR10 .... 4 track
Otari MTR10 .... 2 track
MCI JH 110 .... 4 track
MCI JH-110B .... 2 track
MCI JH-114 .... 16/24 track
Technics RS 1520 .... 2 track
Technics SV-100 digital .... 2 track
Sony PCM-F1 digital .... 2 track
Studer-Revox B710 cassette

Console:
MCI JH536 automated
MCI JH636

Power Amps:
Crown Delta-Omega 2000
Assorted Crown, Crest, BGW, and Yamaha amps

Monitor Speakers:
UREI 813B
Eastlake TM-7
Tannoy SRM-12B
Auratone 5C
JBL 4333
JBL 4401
JBL 4673
EV Interface I

Echo, Reverb, Delay Systems:
Lexicon 224 digital reverb and PCM 41 digital delay
Eventide H910 and H949 harmonizers
Sequential Circuits Pro-FX programmable digital delay, reverb, chorus-flanger
BAE LP140 plate reverb and live chambers

Microphones:
Neumann, AKG, EV, Sennheiser, Beyer, Crown, Sony, Shure, RCA
vintage tube mics by Neumann, AKG

Video Film:
Audio Kinetics Q-lock 3.10 synchronizer
MTM 16/35mm high speed projectors
MTM dubbers and recorders
JVC CR-8250-U and CR-8200-U ¾″ video recorders
MCI 1″ layback recorder
Nagra 4.2L location recorder
Sony CVM 2560 video monitor
NEC 45″ video projection monitor

Figure 3. Studio A—live area. Note the diffusive and absorptive elements on the ceiling.

Additional video facilities to come would potentially increase the problem. Therefore, the electrical system, the conduit system, and the interconnect cabling system had to be completely decoupled from the building ground and protected with a full Faraday shield. Otherwise, the metal roof, which might make an excellent grounding source for a future uplink installation, could transmit RF interference through the electrical system...and potentially through all the audio systems!

Two technical grounds were drilled at Dallas Sound Lab to solve the problem: one as a "third wire" for the electrical system, and the other as a separate ground for the audio cabling system.

The roof deck measures 26 feet in the clear from bedrock to bar joists. A concrete slab, with concrete pillars to bedrock, includes metal posts mounted on the pillars to support a metal pan roof deck. Three inches of lightweight concrete lie beneath an additional three inches of buildup roofing and gravel. Extremely lightweight aerated concrete posed the challenge of extremely low transmission loss, placing exceptional requirements on the ceiling cap. Three layers of ⅝-inch gypsum board, sealed to the interior side walls, four feet under the roof deck, proved to be the solution.

After the floating concrete floors were poured, perimeter walls were erected. Then interior building walls were erected on top of the floors. All these walls went from slab to roof deck, and were sealed all the way up, even around the bar joists. Then floating ceilings were suspended from the bar joists on rubber and spring isolators. Heavy doors, sound rated at STC 48, were installed throughout the facility.

STUDIO A

The primary visual requirement was that the design provide a good view of the control room into the studio area including the projection screen (see FIGURE 2). Large orchestra and rhythm sections should be recordable simultaneously; diffused live sound in the main area should have a relatively long decay time.

Midband frequency decay time measures .9 seconds, and is flat with frequency up to approximately 8 kHz, continuing flat down to 100 Hz. An apparent rising decay, created through mic'ing techniques and modal response, provides warmth and sonority to the instruments, as if in a concert hall setting.

Good reflections back into the orchestra area furnish a heightened sense of ensemble among the musicians. This approach, according to Berger, provides a pleasant contrast to the difficult time most players have in many anechoic, or dead, recording studios. Therefore, much time was spent creating a reflection path in the room to create a diffuse field with an even decay over the full frequency spectrum for good multiple mic'ing technique (see FIGURE 3).

The room was shaped so that reflection patterns would simulate an Initial Time Delay gap (ITD) of a much larger room. This is, in effect, the ratio of the path between the direct sound and the microphone, and the direct sound and its reflection.

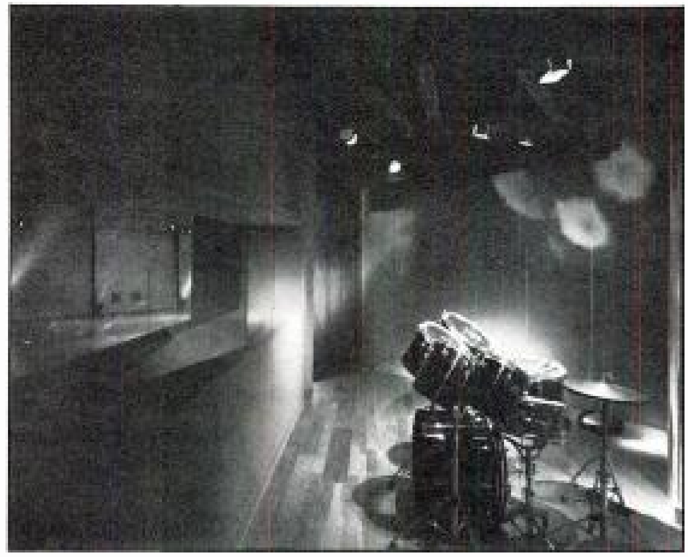


Figure 4. The drum booth. Note the angled front glass and, again, the diffusive and absorptive elements on the walls and ceiling.

A three-channel speaker system is mounted behind the projection screen for playback of film scoring sessions. Resiliently mounted on a support truss, each channel is a JBL 15-inch woofer with a constant directivity horn. They are flush mounted and sealed into a wall construction of a 2- by 6-inch wood stud frame, 12 inches on center, covered with two layers of ⅝-inch gypsum board. Four inches of absorption was then applied to the face of the gypsum board behind the suspended projection screen. This provides a massive area for low frequency sounds from the speaker to project into the room.

Adjacent to the live area is a rhythm track area that is more absorptive. Designed for brass, vocals, and guitars, it utilizes the same hard rubber floor as the main studio. Trapping is achieved in the ceiling and side walls. For mic'ing flexibility, the rhythm area opens into the live area. This allows close and distant mic'ing to occur simultaneously in the separate regions. High ceilings enable the instruments to speak into a large-volume, low-impedance path. They are, in effect, allowed to breathe.

THE DRUM BOOTH

Looking into the rhythm area, the booth, an unusually large one (see FIGURE 4), measures 26 feet across by 16 feet deep by 14 feet high; a floating concrete base 16 inches thick is topped by a hardwood floor. Absorptive materials cover alternate panels at the rear. The front wall contains a frequency-selective trap for the kick drum to absorb splash from the snare drum and toms. Further low frequency and midband absorption is provided by a suspended hanging bat system. The hardwood floor creates a short path from source to microphone, resulting in a thick meaty sound.

The sloped cap ceiling above the hanging bat system reflects diffused sound back into the microphones with a delay of 14 milliseconds. This puts an extra ambience around the drums and gives them a chance to breathe.

THE VOCAL BOOTH

Equipped with a similar hanging bat system and absorptive back wall, the vocal booth looks onto the live studio area through 15-foot-wide sliding glass doors. Made of ½-inch laminated glass, the doors are weathered and air-infiltration rated. Talent can easily see the projection screen and the rest of the studio from here.

Though the 9-foot Steinway concert grand piano normally faces into its own absorptive trap in the rhythm area, it can readily be moved into this booth, which offers excellent isolation.

CONTROL ROOM A

The main control room measures approximately 20 feet wide by 22 feet long by 14 feet high—on average, of course, because no walls are parallel. It is clearly designed to hold a large number of people.

The inner front panes of the control room window are sound-rated ¾-inch laminated glass. "Any thinner," said Berger, "and the glass would resonate more than our performance goals would allow us to accept." The monitors, resiliently mounted and isolated on spring and rubber mounts, are decoupled from the interior shell walls and ceiling cap, and are tied directly to the roof structure 26 feet up. This effectively isolates the room from mechanical vibrations of the speaker.

It is a certified LEDE™ (Live End/Dead End) control room. On the front portion of the room, several layers of glass fiber board are stacked in a sequence of increasing density. Sound penetrates a one-inch, one-pound density outer layer, then a two-inch, three-pound density middle layer, and finally is absorbed into a two-inch, six-pound density inner layer. This represents a smooth gradual change of impedance to widely varying degrees of incident sound.

Figure 5. The rear wall of Control Room A.

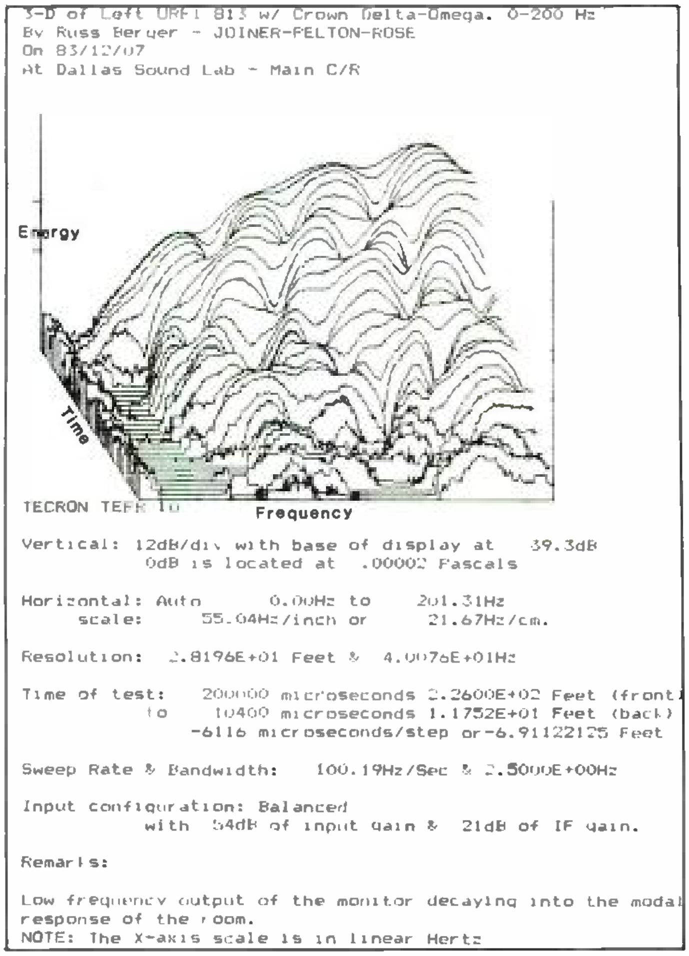


Figure 6. TEF response for Control Room A.

The rear section of the room (see FIGURE 5) is constructed out of multiple layers of gypsum, scored and bent into polycylindrical diffusers and angled panels. They are positioned so that geometric frequencies will be reflected back into the mixing position over a wide variety of time periods, thus creating the desired temporal patterns and ambient field in the room.

Dallas Sound Lab recently hosted the Syn-Aud-Con graduate seminar (sponsored by Synergetic Audio Concepts), an LEDE™ & Studio Design Workshop, where measurements and performance verification of the studio and control room were validated (see FIGURE 6).

The control room floor is floated independently from the rest of the building's structure. A concrete slab rests on resilient isolators, with interior walls located on that independent base.

SUMMARY

"This has been a very ambitious project," said Berger. "Marcon Construction Company and Dring Engineering showed real commitment to quality. It was their first studio project of this magnitude, and they took extraordinary pride and interest in their work. It was a real joy working with Johnny Marshall and Russell Whitaker. They had the dedication, the budget, and the talent to put together a fine studio." ■

BOB ANTHONY

Band-Aid Repairs on the Road (Or How to Eliminate Major Audio Surgery)

A little preventive medicine could spell the difference between a successful tour and a disaster.

FOR MOST SOUND reinforcement companies, the question of providing adequate back-up services in the event of system failure can be both cost prohibitive and expensive. Large amounts of extra gear require substantial investments in capital that could best be used somewhere else. Yet cancelling a show, because something can't be replaced or repaired, means that everybody loses money.

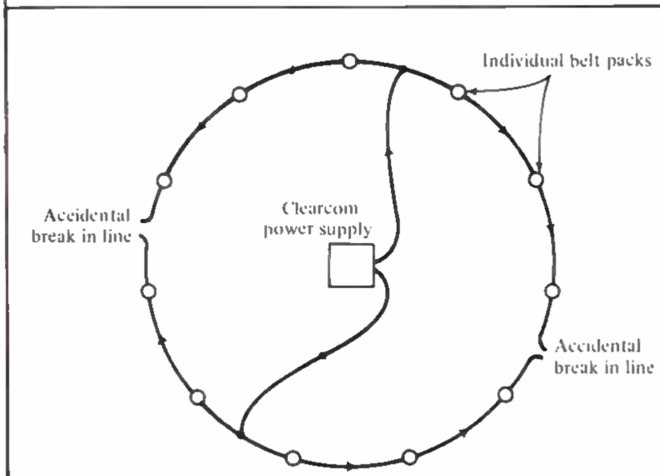
One company that has overcome this apparent Catch 22 situation is A-1 Audio, based in Hollywood, California, whose list of clients include the Doobie Brothers (when they were touring), Barry Manilow, Frank Sinatra, major corporate industrial shows, and a large percentage of the principal hotels and casinos in Las Vegas, Lake Tahoe, and Atlantic City. A-1's approach to on-the-road system maintenance, termed "band-aid service," allows the road crew to keep the system up and running with a minimum of effort and virtually no downtime.

"Very seldom do any drastic electronic failures occur," says Greg Oshiro, the electrical engineer in charge of A-1's Technical Services department. "The vast majority of repairs are quite simple—broken connectors, a bare wire, a loose screw, etc."

A-1 owes this reputation for reliability to their extensive planning, cost-effective in-house equipment modifications, and a unique sound-system design that takes into account the rigors of the road as well as the sophisticated listening demands of contemporary music fans. "We've tried to build a bullet-proof, fail-safe system," says Al Siniscal, president and founder of A-1 Audio.

Bob Anthony is a free-lance audio writer working out of the Los Angles area.

Closed-Loop Communications



A faulty communications system can spell the difference between a good show and one that just doesn't synchronize. For small shows with only a few stations, A-1's intercom system is run through the standard microphone snake cable. If something goes wrong, they just move the signal over to another line. But for shows utilizing over a dozen or more stations, such as complex industrial presentations or huge concerts, the company turns to closed-loop communications (see diagram of wiring scheme). If a break—or even two—occurs, all the stations are still connected to the loop and fully operational.

Field Maintenance Case

TOOLS
Handtools
Power Screwdriver and bits
Soldering iron and stand
Solder
Solder wick and solder sucker
Bench vise
Flashlights
Special insertion and removal tools for multi-pin connectors
Electric drill and bits
Carpentry tools

CONSOLE PARTS
Spare input module
Spare output module
Spare electronic parts (ICs, capacitors, pots, transistors, etc.)

HOIST PARTS
Spare control connectors
Spare contactors
Spare control microswitches

CONNECTOR PARTS
Spare pins and sockets (50-pair)
Spare connector blocks
Spare guide pins and sockets
Spare receptacles
Spare hardware
Spare XLR connectors
Spare ¼-inch connectors
Spare speaker connectors

TRANSDUCERS
Tweeter diaphrams
Midrange diaphragms
Woofers
Small speakers for personal monitors

AC CONNECTORS
Spare 30A 125/250V twist-lock male and female, chassis and cable
Spare 15A 125V twist lock male and female, chassis and cable
Spare 15A 125V Edison, male and female cable
Spare 15A 125V twist-lock receptacles
Spare 15A 125V Edison

TEST EQUIPMENT
Cable checkers for XLR and A-1 specials (VIP, 8-pair, speaker cables)
Digital multimeter
Phase checker
Oscilloscope
Sweep generator

OTHER
Adapters
Signal ground-lifts
AC ground-lifts
Outlet testers
Spare casters
Nuts, bolts, etc.
Mic cable
Speaker cable
Hookup wire
Wiremarkers
Fuses
Meter bulbs
Littlelite Bulbs
Audio tranformers
Miniature toggle switches
Contact cleaner
Electrical tape
Gaffer's tape
Wire ties
Batteries
Velcro
Tie line
Mic clips
Mic stand repair parts
BGW 750 modules
BGW 750 relay cards
Service manuals

GENERAL PREPARATIONS

Training is the first concern. Without a competent staff, even the best equipment is worthless. A-1 takes full advantage of the manufacturer-supported workshops, such as those sponsored by JBL (re-coning speakers) and Columbus-McKinnon (Loadstar hoist maintenance), to provide valuable training for crew members both in the shop and on the road. "At least one person on every flying tour has gone through the Loadstar school," says Siniscal, "so they know all about the repair and operation of the chain hoists."

To augment these instruction programs, A-1 regularly holds their own internal training sessions on topics such as power distribution, equipment interfacing, grounding and shielding, the use of test equipment, general audio electronics, and specific equipment operation. The company also requires every mixing engineer to spend some time working in the technical shop. Conversely, the permanent technicians in the shop must have had some first-hand road experience to understand the importance of equipment that works correctly.

DEPENDABLE DESIGN

To achieve a system that sounds good and is easy to repair, A-1 Audio chose to design their own speaker cabinets—the seven-foot-high VIP Series that houses the power amps and a complete set of transducers in one self-contained cluster. Although many companies frown on this approach, A-1 has found myriad advantages.

The company chooses only high-quality, rugged and reliable gear, such as the BGW 750B power amplifiers. That particular model is modular, so parts may be swapped quickly with little or no technical equipment or

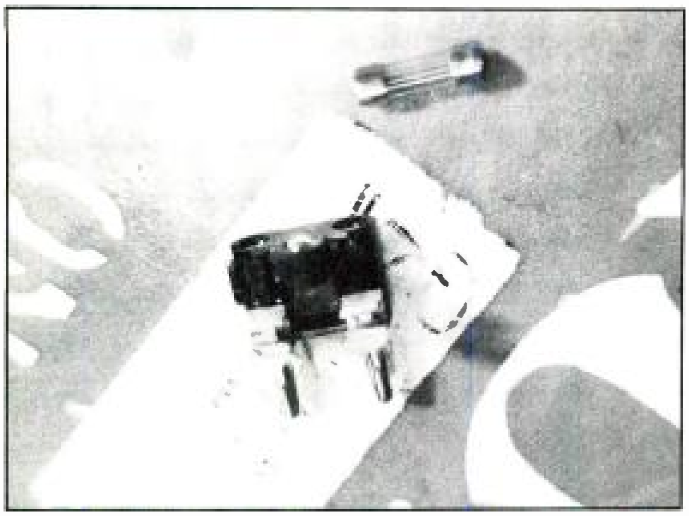


Figure 1. Banana plug with fuse for speaker protection.

knowledge. This allows A-1 to limit their back-up supply to one or two replacement modules and a couple of spare amps per tour. The defective board is then sent back to the shop where the repairs are completed. "We don't expect people on the road to exchange transistors and capacitors." Greg Oshiro explains. "We'd rather have them unplug a module, replace it with a new one, and get back on the air. We'll worry about why the transistor blew when the module is back in the shop and we have the time to do tests and measurements."

But breakdowns rarely occur. The company's philosophy is to avoid limiting at the power amp in order to preserve as much dynamic range in the system as possible. They utilize amplifiers that can produce more signal level than the speakers require before the onset of clipping, which is liable to overheat and generate more amp failures. According to Siniscal, "The reserve power capability for this combination of components allows the BGW's to run much cooler and to cleanly reproduce a signal that contains a lot of extremely high-level, short duration transients and repetitive low-frequency sounds. The amp is just coasting all the time, which eliminates clipping and blowing out amps or speakers."

If, for some reason, an amp does go down, their close proximity to the speakers makes it easy to double up the transducers on the still-working channels via a jumper cable. In fact, a whole cluster could go up in flames and the rest of the system would still operate properly with all audience areas adequately covered with sound. Cluster dispersion patterns are overlapped so that two adjacent clusters can cover the loss of the cluster positioned between them.

Separate AC runs are fed to each cabinet through either one of two AC connectors mounted on the back panel. Both are capable of powering all the amplifiers for that particular cluster, but the unused receptacle automatically shuts off via a relay powered by the functioning AC connector. In the event that one AC input is defective, the second is immediately accessible. (The power connectors are of the twist-lock type to ensure that power cables don't vibrate or pull loose during the show.)

With the amps and speakers mounted in the same boxes, speaker lines never get out-of-phase or mispatched, because they are never unplugged. The entire cluster and flying truss is disconnected and rolled into the truck for transporting to the next show. The only audio connections that need to be made at each installation are the input lines. These are run through a single 11-pair snake cable outfitted with a military-type, bayonet connector to ensure a very quick, positive connection and to eliminate the possibility of stripped threads.

A-1's touring systems are designed to cover three zones in a typical concert situation. The floor area of the arena is designated as Zone 1; Zone 2 usually comprises the seats in the central back portion of the arena, and Zone 3 is comprised of the seats on both sides. Each zone receives three signals—high, mid, and low—balanced specifically for the particular section. A fourth, full-range signal is available for a rear-fill system in the back. The eleventh line is held in reserve as a replacement, should one of the others fail.

"By having the three zones on one multi-pin connector, we can never get the lines mixed up—the lows plugged into the highs, the highs into the mids, or whatever, says Siniscal. All the crew has to do is plug in one connector and all the lines are connected correctly. A three-way switch on the back of the cabinet selects which set of lines are hot for that particular speaker location. If one of the sets should fail, the engineer simply turns the switch to another zone setting and adjusts the volume on the amp. In an extremely unusual case, where all nine lines are unusable, there are still three options open—use the single full-range line feed, the spare line in the cable, or attach jumper cables from an adjacent cluster."

If for any reason a line of the drive snake goes bad or develops a hum or buzz during a performance, the entire show can be run on only two zones, or one if necessary. Three graphic equalizers and three electronic crossovers at the house console position allows the engineer the flexibility to adjust the zone levels right from there.

ADDITIONAL SPEAKER PROTECTION

To further protect the speakers, A-1 has installed fast-acting fuses between the amps and speakers (see FIGURE 1). Located at the back of the cabinet, in-line between the output of the amplifiers and the input of the speakers, is a banana-plug connector that can be used as a test point for those components. Dual banana plugs are used as jumpers between both the plus side and the minus side to each speaker, with the minus jumper being hardwired across the prongs. The plus jumper, however, has a standard fuse holder installed on the plug through the little holes where the set screws normally go. A fast-acting, low-resistance fuse prevents damaging signals from reaching the speakers. In addition, a tiny, 48-volt incandescent lamp fits snugly into the round wire opening on the side of the standard MDP banana plug. If the fuse does blow, the bulb lights up, thus indicating which cabinet has malfunctioned.

Al Siniscal relates one such instance where the fuses saved the entire system, the show, *and* a bundle of money. "During rehearsal at an outdoor concert, a keyboard player for the opening act decided to throw a switch on his electric piano from Mic Out to Line Out, because he thought he wasn't being heard in the house. At the same time he was holding down the sustain pedal with a chord and a bass note going through both our system (on the towers) and the band's system (on stage). Needless to say, when he threw the switch the transients were so severe that they blew half of our protection fuses despite the fact that we had limiters in-line."

Fortunately, A-1 needed only to replace some fuses and their system was back up and running. All the speaker coils for the opening act's sound equipment, however, were destroyed, which meant retiring their system for the remainder of the tour. But even if the band had carried enough spares, several hours would have been required to take out all the screws, remove the backs of the cabinets, unbolt the speakers and the connectors, and finish the repairs. "After the promoter found out what happened," remembers Siniscal, "he came over and told me that we saved him over $150,000 dollars by not having to cancel the show."

SUB-SONIC FILTERS

While some companies strive to produce sound systems that are close to flat frequency response down to 5 Hz, very little useful audio information exists below 25 or 30 Hz. A-1 Audio hired Bill Isenberg, the designer of JBL's 5234 crossover, to develop a 25 Hz filter card (18 dB/octave; down 3 dB at 25 Hz) that could be added to the device. According to Greg Oshiro: "If the signal isn't rolled off around that point, the woofer has to make a lot of unnecessary excursions. Before installing the filters, we used to lose (on the average for a large arena tour) about one woofer a day. The filter has reduced that number to maybe one speaker every two months. Plus, the overall sound quality of the system is better, because the woofers are no longer generating 5-cycle notes, which just muddy the system and waste power."

PRE-ROAD MAINTENANCE

To ensure that there are no surprises once the tour starts, the technical department spends a great deal of time on pre-road maintenance. "About 90 percent of the breakdowns we see," says Greg Oshiro, "are simple mechanical failures, because we've developed check-out procedures to catch major potential failures before the gear goes out the front door."

For example: A bi-amp rack containing two stereo amps, one crossover, a harness, and a panel takes about three to four hours to check out. The amps are completely disassembled and tested module by module for distortion (at frequencies of 20 Hz to 20 kHz), noise, frequency response, shock, solid connections, etc. Each piece (two amps, crossover, harness, and the rack itself) that successfully passes the test receives a green tag with the date and the technician's initials to verify that it was serviced.

EXTRA CONSOLES

Consoles very seldom break down completely, but rather than using one house console with 48 inputs for a big show, like the recent Barry Manilow international tour, A-1 Audio sends out two boards—a 32-input Harrison Alive console and a matching 24-input Harrison Alive board—with an interconnect system between the two. If a board is damaged or goes down, the engineer can still do the show on just the remaining board. "He may not have as much versatility with only one board," says Siniscal, "but neither will the promoter have to send 20,000 people out the door. That would be disastrous. Although that's never happened to us, the double-board concept provides us with an extra margin of safety."

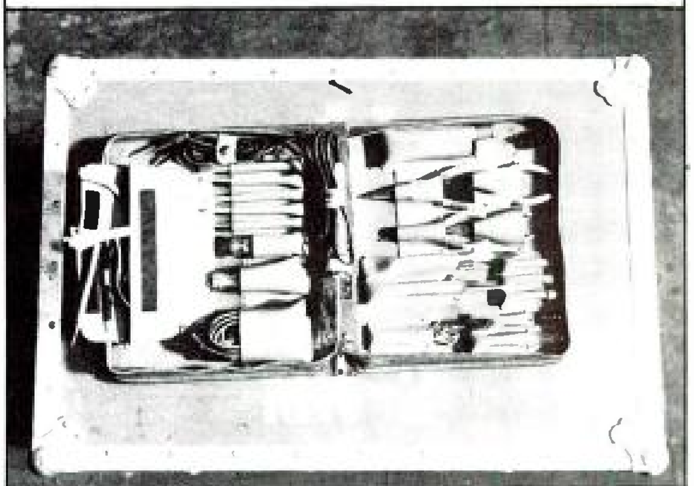
As added insurance, every tour travels with spare input and output modules, as well as two power supplies per board. Siniscal figures that "at the most, we may lose only one power supply a year in all the shows we do. But if something does happen, we know we're covered."

SNAKE CABLE

A-1 Audio custom-builds their own 50-pair snake systems. The cable actually contains 51 pairs of individually shielded and jacketed pairs, thus providing a spare line. Each twisted pair is 22 AWG with an aluminized Mylar foil shield and drainware inside a nylon jacket. The outer coating is comprised of a nylon-fiber spiral wrap and an extra-thick Neoprene rubber jacket. The end result resembles 51 separate microphone lines inserted in a single outer covering that looks like a heavy-duty AC power line. "We find the audience doesn't trample on it, because they think it contains a dangerously high voltage," notes Al Siniscal. "That kind of cable is heavier to carry in and out, but it's a lot more reliable. If someone steps on it, the pairs don't rub together and short out, because the individual jacket on each one of the pairs provides the extra protection."

On the ends of the cable are 50-pair AMP connectors of rugged GE Lexan. If a plug breaks, the various

Personal Tool Kit



Rick Southern's customized Jensen tool kit

As Greg Oshiro, Director of Technical Services, points out: "The most valuable piece of test gear that goes on the road with every system is the one guy who's in charge of maintaining the system. A good maintenance engineer, a set of hand tools, and a multimeter, can fix almost any failure." Audio engineer Rick Southern, who has mixed shows for artists as diverse as the Doobie Brothers, Frank Sinatra, McCoo and Davis, and Barry Manilow, also does his own repairs when necessary. Here is a compact tool kit (model #JTK-16) manufactured by Jensen Tools, Inc. (Tempe, Arizona) that he has customized to his own needs:

A set of phillips-head and flat-head screwdrivers
Several needle-nose pliers
A Shure tone generator A15TG
Assorted wrenches
X-acto knife and extra blades
Small ruler
Three sets of test probes
Soldering iron and solder
A set of hex-nut wrenches
A set of Allen wrenches
VOM

components may be replaced without time-consuming soldering. The pins pop out with a removal tool.

COMPUTERIZED INVENTORY FOR EQUIPMENT CONTROL

How many times have you gotten out on the road and discovered that a very important tool or electronic component was nowhere to be found because it wasn't shipped or went out with the wrong tour? A-1 has eliminated that situation by color-coding the cases of each system with colored gaffer's tape (to avoid confusion and mixups during system assignment and storage) and computerizing their entire inventory system.

With the cost of microcomputer systems dropping by the week, the thoroughness of a computerized inventory maintenance system saves money, time, and frustration. Every piece of equipment that contains a serial number is bar-coded and entered into the computer by brand name, model number, serial number, weight, cost, and several other applicable parameters. As cases are packed for a tour, the piece of equipment is read with a bar-code reader. Likewise, packed cases are also read as they leave the warehouse for the truck. The information is then assigned to the client's file and an equipment manifest is printed out containing a complete list (International Guarantee Chain Carnet sheet) of equipment, total weight, total dollar value, equipment volume in cubic feet, the number of the shipping case in which the equipment is packed, country of origin, and several other necessary values. Bob Marshall, the man in charge of Computer Services at A-1, explains: "The computer system ensures us that everything is where it should be, and we have the parts and tools we need to make the show run without a hitch. And during check-in, the same bar-code system verifies that we get everything back."

ON THE ROAD

Once the equipment is out the front door, the responsibility for keeping the system operational is turned over to the road crew. To facilitate on-the-road repairs, each tour travels with a case of test and maintenance equipment. Al Siniscal emphasizes that: [A-1] spends a lot of money on those kits, because we feel it's necessary to maintain the gear as much as we can as we go along rather than waiting until the equipment returns to the shop. Why travel around with pieces broken or slightly malfunctioning? If something does happen, we can replace or repair the various parts."

In addition, A-1 Audio provides the road engineers with a dedicated office containing Watts lines, desks, and all the proper equipment and conveniences to make preparations for tours. Engineers also have access to a library of files specifying floor layouts of all the venues the company's clients normally play in. "Before they even get to the venue, the engineers exchange a lot of information that can help them make the right preparations and eliminate the headaches of being caught off-guard," says Siniscal.

"Band-aid service" is possible only after the time and effort has been invested in planning ahead and preparing for the worst, such as the trauma of several blown speakers or a dead console. Adequate forethought is the key that reduces potentially expensive catastrophes to minor inconveniences. And even the smallest company can afford to plan ahead. ■

JOE COENCAS

How to Reduce the Risks of Producing On-location Events

FIFTEEN YEARS AGO, the Rolling Stones performed at the Altamont Speedway outside San Francisco and the frenzy resulted in chaos and murder. Last summer, a freak electrical storm aborted Diana Ross's concert on the Great Lawn in Central Park. And in Washington, D.C., the heavily publicized Fourth of July Wayne Newton concert was almost rained-out, delaying the performance.

Thankfully, disasters such as these are quite rare; hundreds of live events, in fact, take place each year with relatively few problems. What the occasional problem event dramatically demonstrates is the extreme potential risks and losses producers face each time they take a show on-location. The nightmare of damages and losses (which one hopes will never come true) can include personal injury to the performer, production crew and/or members of the audience, the loss or damage of valuable equipment or the non-appearance of a key performer.

Fortunately, however, specialized insurance protection designed to protect against calamities such as these does exist. It is imperative, therefore, that producers, promoters, broadcast groups, sound reinforcement companies and equipment rental houses involved in the increasing number of live-event productions be aware of their operating risks and exposures and become familiar with the kinds of insurance policies available. Only then will they be able to reduce the risks and financial losses that may be incurred in such ventures.

For the layman, the most commonly held misconception is that all production insurance policies are alike and that these policies are so detailed, complex and laden with loopholes that only an expert is equipped to interpret what's covered and what's not. Yet if one takes the time to investigate the handful of reputable insurance companies providing coverage for entertainment-related projects, the dreaded fog soon disappears.

Chubb Inland Marine, a division of the Chubb Group of Insurance Companies, has written coverages for the telecommunications and entertainment industries from the days of the hand-cranked phone right up through our present day satellite networks, including live broadcast events and rock concerts. For the sake of clarity, the insurance described in this article will be detailed according to Chubb insurance standards, though similar types of coverages may be available from other insurance companies.

THE AUDIO-VISUAL PROPERTY FLOATER

The "Audio-Visual Property Floater," as the coverage is commonly known, is an all-inclusive insurance plan designed to cover, on a very broad basis, the most common incidents that can result in direct physical damage or loss of property that is scheduled under the policy. Whether it be a professional musician playing in a string quartet, a major symphony orchestra, a touring rock band, a sports competition or a political convention, the policy provides "all-risk" coverage for the insured's instruments, sound reinforcement and lighting equipment, wardrobe, props and sets, sheet music or any related property while stored in permanent premises, in transit or on temporary location anywhere in the world. Each A-V Property Floater policy is custom manuscripted to suit the specific needs of the client.

"To afford proper coverage," says Alma Bailey, manager of Chubb's Inland Marine Department, musical instruments and production equipment should be valued at the price purchased (including taxes) and/or its current market value. Appreciation of all equipment or one of a kind items can also be taken into account." Ms. Bailey continues: "Very often, for example, expensive privately owned or leased mobile vans crammed with hundreds of thousands of dollars worth of production and post-production videotape and/or film equipment which accompany the tour are given priority consideration. And the 'high risk' hazards of weather damage, vandalism and accidents that may result from long hours of travel also are considered when the insurance contract is drawn."

Theft, Ms. Bailey emphasizes, is by far the major peril faced by today's insureds. She also notes that with the increasing numbers of performers traveling to far corners of the world, government confiscation of property is an overlooked exposure.

Ms. Bailey adds that different types of coverage options exist for the production company that owns its own equipment, as compared with equipment rental houses which have far less control over how their hardware is being used and cared for by the renting musicians or production company.

ADDITIONAL INSURANCE COVERAGES

There are, however, many other perils that may plague a production in addition to loss or damage of property. Most often, these risk exposures fall into three categories: General Liability, Non-Appearance and Business Interruption Insurance.

General Liability Insurance is a comprehensive policy designed to protect the insured against claims of negligence. Like the homeowner who obtains this kind of insurance to protect himself against the person who sues because he slipped on some ice on the walk outside the insured's home, promoters of live events must have this coverage to offset claims of bodily injury or negligence or damage to other people's or their property.

In tragic cases, like the murders at the Altamont Speedway or the deaths of several fans who were trampled by overly eager fans struggling to get into a Who concert, it is important to note that it is the promoters, not the performers, production companies or broadcast companies who are responsible.

Non-Appearance Insurance is designed to cover the promoters' up-front expenses in the event an entertainer, guest speaker or key personality fails to appear for the event. Needless to say, the value of such a policy is immeasurable to those producers risking the loss of hundreds of thousands of dollars of revenues incurred because of cancelled performances and/or returned tickets. This policy which responds to actual losses, however, is not to be confused with Business Interruption Insurance.

This Business Interruption coverage is customized to protect the insured against loss of revenues resulting from mishaps that can alter, delay or stop the show. If, for example, an amplifier, camera, large projection screen or some other key pieces of equipment gets lost or damaged in transit, the program may be delayed until the item is either replaced or repaired. For such an incident, the production company would normally have to incur the resulting additional costs. With Business Interruption coverage, however, the insurance company would respond to the financial losses assumed by a cancelled or postponed performance.

NEED FOR COVERAGE

There is no question that proper insurance coverage is an absolute necessity for any individual or company planning to produce a program or concert on location. Since there are relatively few insurance firms involved in the business and only a handful of different policies to choose from, investigating the kinds of coverage that best suits one's needs need not be an intimidating, confusing experience. More important, as the industry changes and progresses to meet audience demands, it is equally important for the insured to review coverage options to obtain the most comprehensive, up-to-date insurance policies available. "The first step should be to choose an insurance company that is reliable and able to respond quickly to your priorities," concludes Ms. Bailey. "A representative will be happy to explain the kinds of policies available and give advice as to which one(s) are most appropriate."

She adds that policies covering tours are usually reviewed annually, so once the insured goes through the initial time-consuming stage of itemizing the value of musical instruments, video and sound equipment, costumes, etc., the agony is over and you can concentrate your efforts on the production at hand. ■

JIM WELLS

Audio Musical Design For Animated Shows

Part II: The Bullwinkle Restaurant Show

Here, we take a look at the process of Aural Image Animation, as author Wells and others bring cartoon characters to three dimensional "life."

IN PART I, we described some of the design criteria necessary to create a free-standing, portable, multi-character, audio/musical (A/M) animated Christmas display for continuous use, specifically in shopping malls. In an effort to simplify setup and handling while still providing true multi-character multi-point source imaging, speaker components were incorporated into, or "on board" each animated figure. By attaching transducers on board moving parts of the figures (in this case the heads), the speakers and aural image also become animated. Moreover, the visual and aural images are physically synchronized.

The creation of Aural Image Animation (A.I.A.) audio systems required a certain amount of re-thinking of studio procedures. First, special signal processing is necessary to compensate for the unusual nature of the enclosure—in this instance, the hollow moulded Latex head of each Christmas elf. Secondly, the complex animated aural images of multi-character ensembles with their breathing (nicknamed the "lighthouse" effect) and subtle doppler shifts (nicknamed the "calliope" effect) simply cannot be referenced to conventional, static studio monitors. They require direct reference to the animated event, or the use of the display as a monitor system in the studio during the mixdown—the L.I.V.E. mixdown.

You may think, as we did in the beginning, that

Jim Wells, president of Fantasonics Engineering, is an audio/musical designer specializing in dimensional animation.

animated figures with built-in speakers are commonplace...but they're not! To understand just why on-board character speaker systems and direct interpretation are innovations, we must dig into the history of dimensional animation.

ROOTS: THE ART AND THE CARTOON

Say "animation" to most people and they visualize a cartoon, but say it to any one of a small group of Dimensional Animation (D.A.) designers, and you stir up visions of microprocessors, electro-pneumatics, hydraulics, fiberoptics, and mechanical fabrications that would make NASA jealous. D.A. has come a long way since the early days of Warner Bros. and Disney, but cartoon traditions still greatly influence the way in which D.A. is created, especially in the areas of program production and audio system designs.

THE CARTOON SOUNDTRACK

The cartoon is conceived in art form; whether it's one frame or 70,000, it must "tell a story." Concept sketches representing each scene and piece of action in real time are pinned to the wall, often stretching around the room.

After storyboard development but before a single frame of the actual cartoon is drawn, a script and score are conceptualized in preparation for soundtrack creation. At this point, methods differ among cartoon animators, but music, dialogue, and sound FX must be performed and recorded—either to a clicktrack or separately—for later assembly.

The last step in the cartoon soundtrack creative process is the mixdown, where dialogue, music, and sound FX are combined using a typical theatre-type speaker system as a reference. Only after the soundtrack is complete may the thousands of individual pictures necessary for animation be drawn and edited to "fit into" the soundtrack, frame by frame.

THE CARTOON SOUND SYSTEM

Design of the theatre audio system was not part of the creative cartoon process, nor were animators able to exert any real influence or control over playback equipment in the field. Still, the movie industry did develop a standard of sorts; theatre systems were generally some combination of horn-loaded bass enclosures and radial or multicell high frequency horns, allowing soundtracks to be prepared in the safe, clinical confines of the recording studio or scoring room equipped with similar or "typical" theatre monitors. Theatre systems were (and are) designed and installed by separate contractors.

THE EVOLUTION INTO THREE DIMENSIONS

Walt Disney was the undisputed innovator in creating art from fantasy, transferring it to the cartoon screen, and then into three dimensional reality. Cartoon-type storyboarding is usually the first step in D.A. design, especially in those theme park attractions and dark rides where people are moved through multiple scenes or events in sequence. Today, a great many of the soundtracks for D.A. are accomplished in much the same way as their cartoon cousins, many times by cartoon, motion picture, or television production scoring groups "moonlighting" in D.A.! Here again, various parts of the soundtrack are assembled and referenced to "flat" monitors in the studio. Often the soundtrack is in the can before an audio engineer and/or sound contractor is called in to install a flat sound system for playback of these programs. Only after the soundtrack and sound system are in place may the animation microprocessor be programmed to move the figures to fit the sound, frame by frame. (Many D.A. microprocessors scan at the same frame rate as their 35mm cartoon ancestors: 24 fps.) The traditional separation of music creation and audio design at the studio reference monitor is evidenced in the types of D.A. audio systems in general application today (there are basically only two).

THEATRE SOUND SYSTEMS

Without a doubt, the type of audio system prevalent in modern D.A. is the theatre system, and there is good reason for this: It works! We are all accustomed (or conditioned) to believing the concert reinforcement listening experience at live events. The movie theatre sound system (from the cartoon) was a natural for D.A.

Specific system configurations may be as varied as the shows they reinforce, but these systems are invariably comprised of theatre- or P.A.-type components applied either to the sides or, in proscenium clusters, above the stage. Program format may be stereo or mono, and as tradition would dictate, is mastered completely in the studio on typical reference monitors. This is also the least expensive approach.

MULTI-POINT SOURCE SOUND SYSTEMS

In some circumstances, it can be anticipated that the viewer/listener will be in more than one position relative to an animated event, such as people moving past, around, or even through an animation. When such a design involves more than one character, character group, or event, each is provided with its own referenced speaker enclosure (usually bookshelf-type monitors) and a discrete voice. In this way, a separate point source is created at each figure location, while collectively, multiple figures build, or present an appropriate relative aural and visual image to many vantage points.

Multi-point source (or dimensional sound) systems are substantially more expensive, requiring multichannel playback and amplification components and the individual development of each speaker location for optimum sound and cosmetic concealment in stage and sets, etc. With the exception of our simplified, inexpensive, and portable Christmas display, multi-point source applications are normally found only in the more elaborate, mega-budgeted fixed installations. This method also provides a substantial improvement in the believability of the aural image over reinforcement or theatre techniques. However, it is still the static

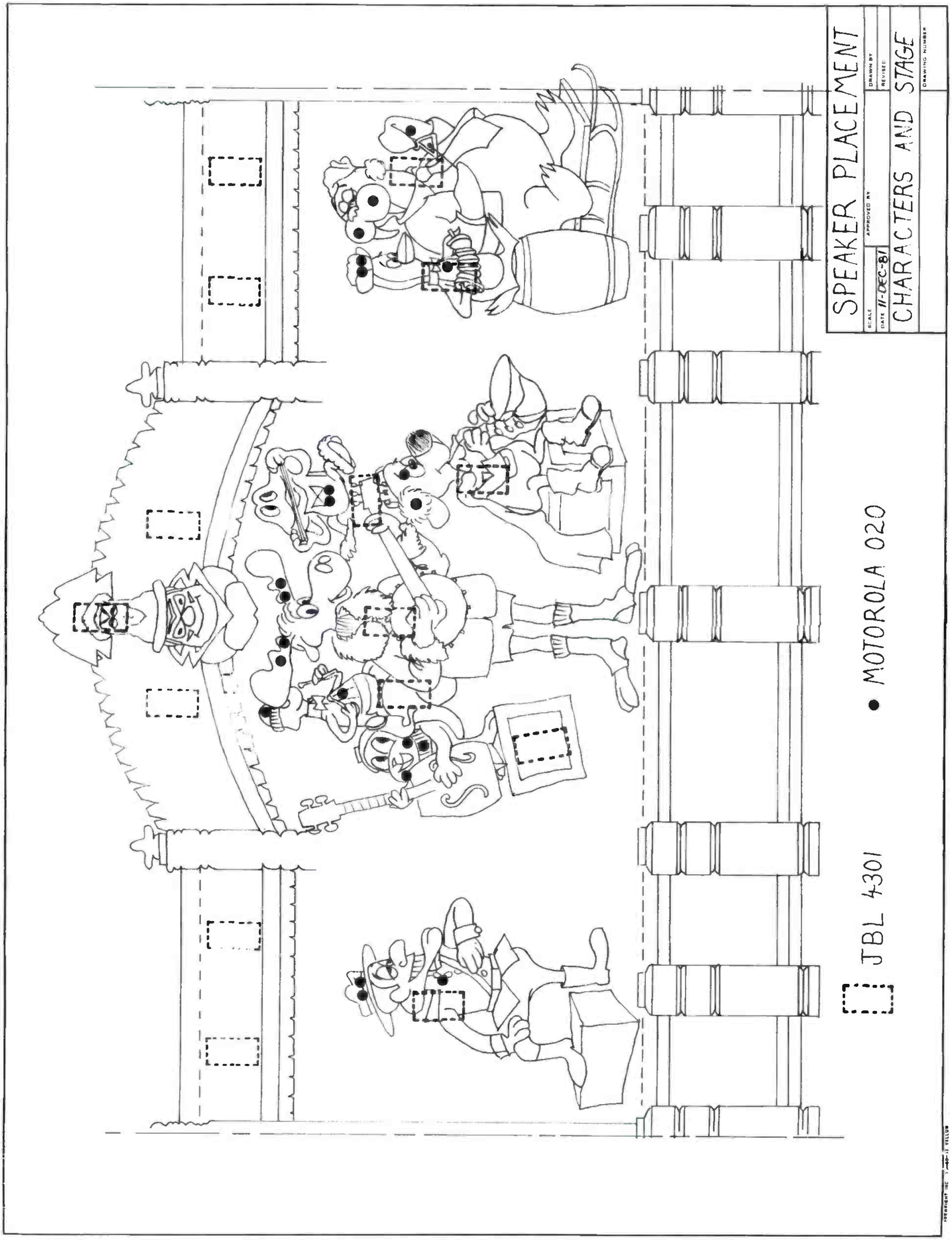


Figure 1. The original art.

application of speaker components that allows preparation of each voice as a monaural signal on similar static monitors in the studio.

WHY FIGHT TRADITION?

In an effort to convey general trends and traditions, I have oversimplified the preceding descriptions. Animators (of all kinds) and their design techniques can be as varied as the shows they create, but the influence of the cartoon is a common thread through both the evolution of D.A. over the past 30 years and the development of each D.A. project. From original art to storyboarding, to separation of the soundtrack and sound system at the reference monitor, to the use of the referenced theatre system (or the use of many referenced systems), each D.A. design is simply the logical extension of the cartoon. So why fight the tradition? If it has always been done that way, there must be a good reason for it... mustn't there?

The Bullwinkle Experiment, or 'The Moose Gets a Soundcheck'

The Bullwinkle Restaurant Show is especially appropriate for this article for several reasons, all of them related to the cartoon connection. First, Bullwinkle and Rocky and the others are established cartoon characters with almost a cult following! This poses an extra challenge to achieve a faithful translation from the T.V. screen to three dimensions. After all, our audience will know exactly how faithful we have remained to the original art.

Secondly, within this single project, we have applied theatre-reinforcement techniques, multi-point source techniques, and aural image animation enhancement techniques, spanning the evolution of D.A. audio system development since the cartoon era.

Last, but not least, the Bullwinkle show is linked to the cartoon by its designers: Fred Hope, Jr. and The Only Animated Display and Design Company, Inc. (or The Only Co., or T.O.C.). There are strong Disney roots in almost everyone involved in this extremely talented design group.

When we were first asked to develop a proposal for the Bullwinkle show audio design, all I could see were antlers over four feet wide—only to me they were semi-horizontal line arrays of on-board smooth analog moves in three planes.[1] The art included in our initial proposal (FIGURE 1) suggested that each character would have a JBL 4301B static enclosure (multi-point source) and a special high frequency on-board array (A.I.A.). Additional speaker enclosures would be mounted above the three stage sections for reinforcement (theatre). Even though the initial animator's specification called for the more elaborate multi-point source reproduction techniques, our proposal to incorporate *any* speakers within the figures themselves was met with a good deal of skepticism. This attitude is not unusual among animators; indeed there is some reason for it.

Keep in mind that the idea to put speakers inside animated figures is not new—it is only sonically successful applications that are new. There is evidence that this *could* be done, suggested in animation patents dating back to the 1930s. However, as recently as the early 1960s, the undisputed masters in the art of D.A. were still unsuccessful in actually accomplishing an on board design.

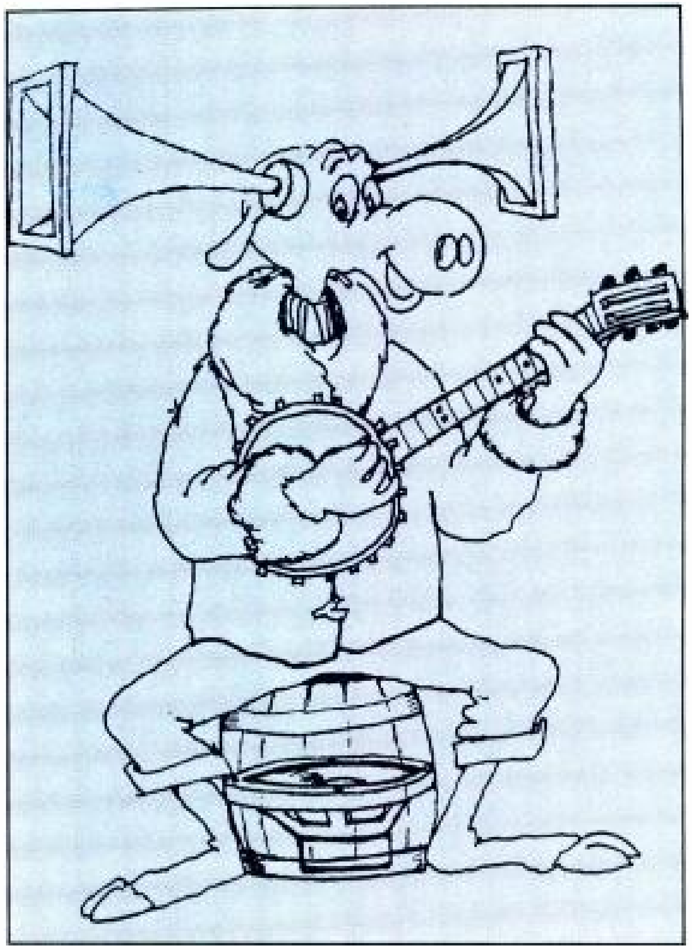


Figure 2. High-frequency horns and barrel-woofer.

A NOT SO 'GREAT MOMENT WITH MR. LINCOLN'

In an effort to create a more believable point-source for Abraham Lincoln in the "Great Moments with Mr. Lincoln" attraction showcased at the New York World's Fair in 1964, Disney animators placed a JBL 2105 5-inch cone in the torso of the President. Problems in the performance ultimately resulted in the disconnection of this speaker, and it was never heard in public.

At the root of those problems was a conventionally recorded, flat voice signal that no doubt sounded great on a studio monitor elsewhere. However, a 5-inch cone mounted in a hollow fiberglass shell is quite another matter altogether. Simply put, backpressure created strong internal resonances, adjacent structures were excited through conduction, and costuming (right down to the authentic 1860s long wool underwear) muffled direct radiation to a point near inaudibility.

In all fairness to the imagineers who developed audio-animatronics during that period, 1964 audio technology probably precluded any real chance to compensate for enclosure characteristics. (Remember when multitrack meant either two or four, and EQ was bass and treble?) They certainly didn't have the capability to manipulate nearly every conceivable characteristic of a waveform that today's digital domain processing affords.

I have the highest regard for the Disney organization, and that's the whole point here...so does just about everybody else! Most animators feel that if Disney can't do it, it can't be done! Hmmm....

MEANWHILE...BACK AT THE UPSIDASIUM MINE

T.O.C. most likely had more practical concerns about allowing audio systems "into" their figures. First, interior space is usually at a premium, especially in the heads (an ideal speaker location), where high concentrations of animation are applied to create expression. Conventional cone drivers with their heavy magnet structures, backpressure, and relatively large size are simply too awkward to fit into many places, and too heavy to attach to extremities where exaggerated movements would produce a more pronounced aural image animation effect.

Then too, animators often keep a certain mystique around their figures: some guard trade secrets to the extent of maintaining restricted access. But even in the most relaxed circumstances, the interior of an animated figure is considered a very private domain—a place where outsiders should knock before entering.

COMIC ART—THE SONIC MODEL

In an effort to break the ice with T.O.C., we appealed first with humorous art (FIGURE 2), and then with a sonic demonstration presented to the key people involved. Keep in mind that innovation is no stranger to this imaginative design group—their skepticism is typical of the entire animation industry.

Our sonic model (nicknamed "the mouthpiece" because he was to argue our case) would have seemed comical if it had not been so completely effective. This demonstration "droid" was simply four Motorola KSN1036A piezo ceramic (PZT) speakers pop-riveted together in a straight line. This was wrapped first in three-inch quilt batting (somewhat acoustically transparent), and then in an outside layer of 1½-inch lime green synthetic material (somewhat less transparent).

Our test signal was the left channel only of a Steely Dan cassette fed directly to the input of a Crown D-60 amplifier in the mono mode. The full-range outputs of this amp were connected in parallel to a JBL 4301B monitor (with the pancake tweeter shut off), and the four PZTs.

The test procedure consisted of turning the volume up, asking everyone to stand back (up to 50 feet in some instances), and listening. We simply animated the aural image by sweeping the model from side to side to create the lighthouse effect for each listener as he came on axis. It really is a lot of fun to watch eyes and faces light up when first experiencing Aural Image Animations in action...hearing is believing.

After first seeing the small size and light weight of the PZTs, T.O.C. gave us the go-ahead to create on-board designs for Bullwinkle and all the rest. The only stipulation (no pun intended) was that absolutely nothing could show on the outside. We were in! The one thing we hadn't mentioned at this point was that by committing to A.I.A.s, we were also committing to a L.I.V.E. mixdown in San Jose, the first store. (When you're breaking tradition, you have to take one step at a time.)

THE AUDIO SYSTEMS FOR BULLWINKLE'S

There are actually three separate audio systems at Bullwinkle's: the stage show, the water show, and an entry or greeting show. The main dining room is over

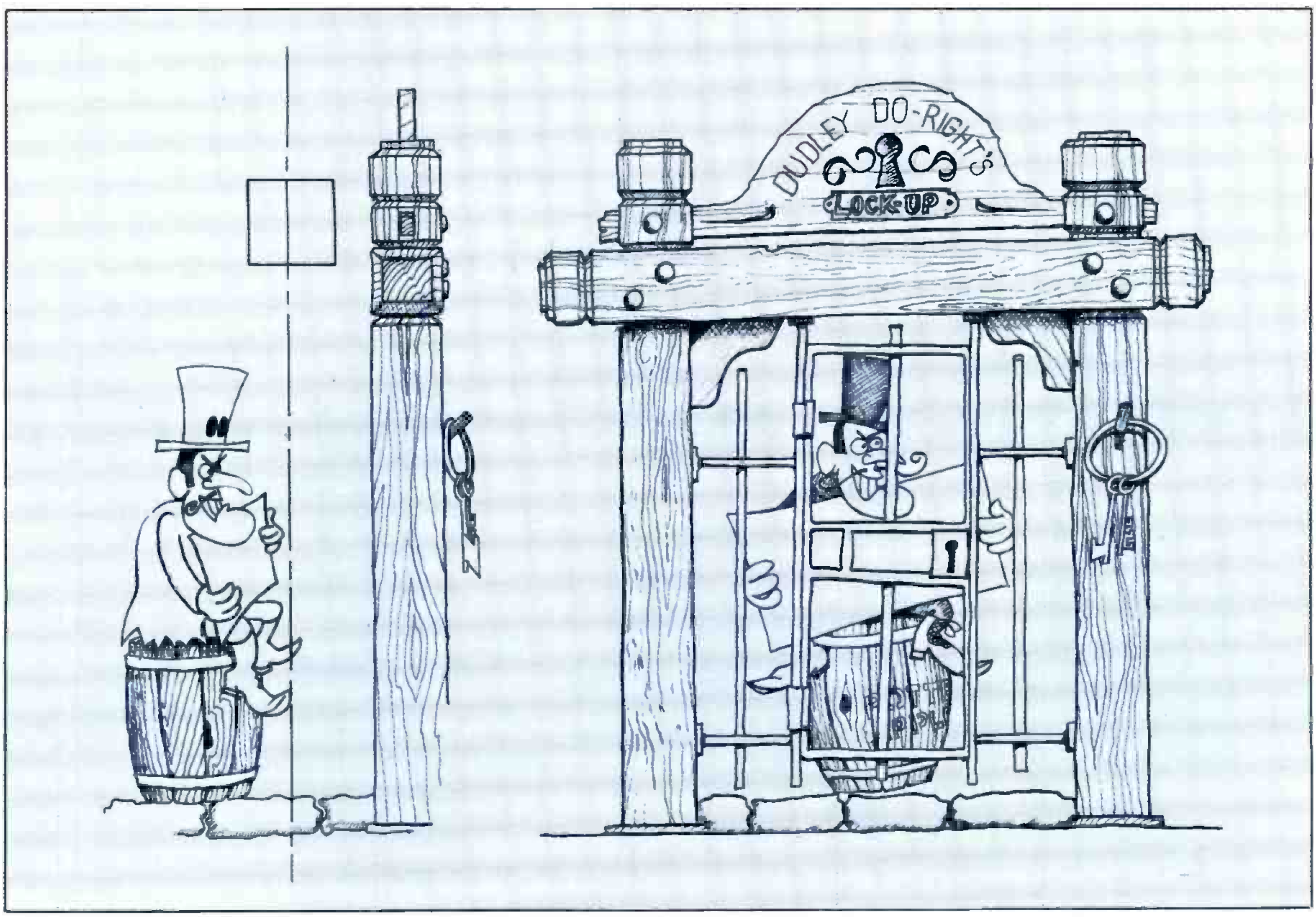


Figure 3. The art should "tell a story."

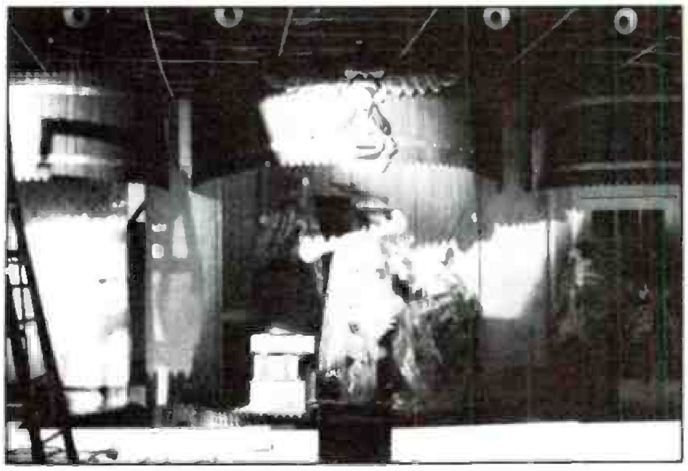


Figure 4. Theatre reinforcement systems "flying" in suspended ceiling. (View from mixing position.)

2,500 square feet, with the animated show stage located in one corner. Directly in front of the stage show is an elaborate and quite animated water show that performs alternately or sometimes in conjunction with the stage show. Upon entry to the restaurant, one is greeted by none other than Snidely Whiplash, who shares a four-track system with a distributed "environmentals" and paging system. We will look briefly at each.

Snidely and the Crickets. Snidely Whiplash is a straightforward audio design. One channel of a McKinsie MMIV four-track continuous loop tape deck is connected to a PS Audio Model II power amplifier that feeds a JBL 4301B enclosure and A.I.A. array, as shown in FIGURE 3. Snidely's movements are not synchronized per se; the microprocessor is also a continuous loop. However, two separate sensing circuits listen to the speaker signal and control his upper and lower teeth independently...it's a great effect.

The remaining three tracks are allocated to environmentals. We did not want elevator music competing with the stage and water shows, yet we couldn't allow dead air between these shows. The motif at Bullwinkle's is Canadian Yukon hunting lodge, right down to the Dudley Do-Right mountie hats and uniforms worn by all employees. A three-track program was created incorporating Canadian northwest woods sounds such as frogs, crickets, birds, etc. A stereo pair was sent to small speakers distributed throughout the complex (even in the parking and approach areas).

The last track is carried by a 600-ohm balanced line to the background music inputs of the paging systems, in synchronization with the stereo pair. The natural voices were assembled around a 5/4 cricket click-track, which proved musical to the extent that we occasionally observed animated fingers and toes between shows.

The Water Show. The water show is the creation of David Usher, who is truly a water sculptor. To appeal to audiences of all ages (important at Bullwinkle's), music programs span the entire spectrum from classical to Star Wars, with water and lights programmed to follow each piece of music in amazingly appropriate ways.

One track of an independent four-track open reel machine contains addressing information for the Microprocessor Animation Control System (MACS), which orchestrates the water show and just about everything else "quite nicely, thank you," (which is what MACS designer, Doug Mobley answers when people ask, "How does it work?"). The other three tracks are left, right, and summed center channel water show program information sent to reinforcement systems "flying" above the three stage segments (FIGURE 4). These modified

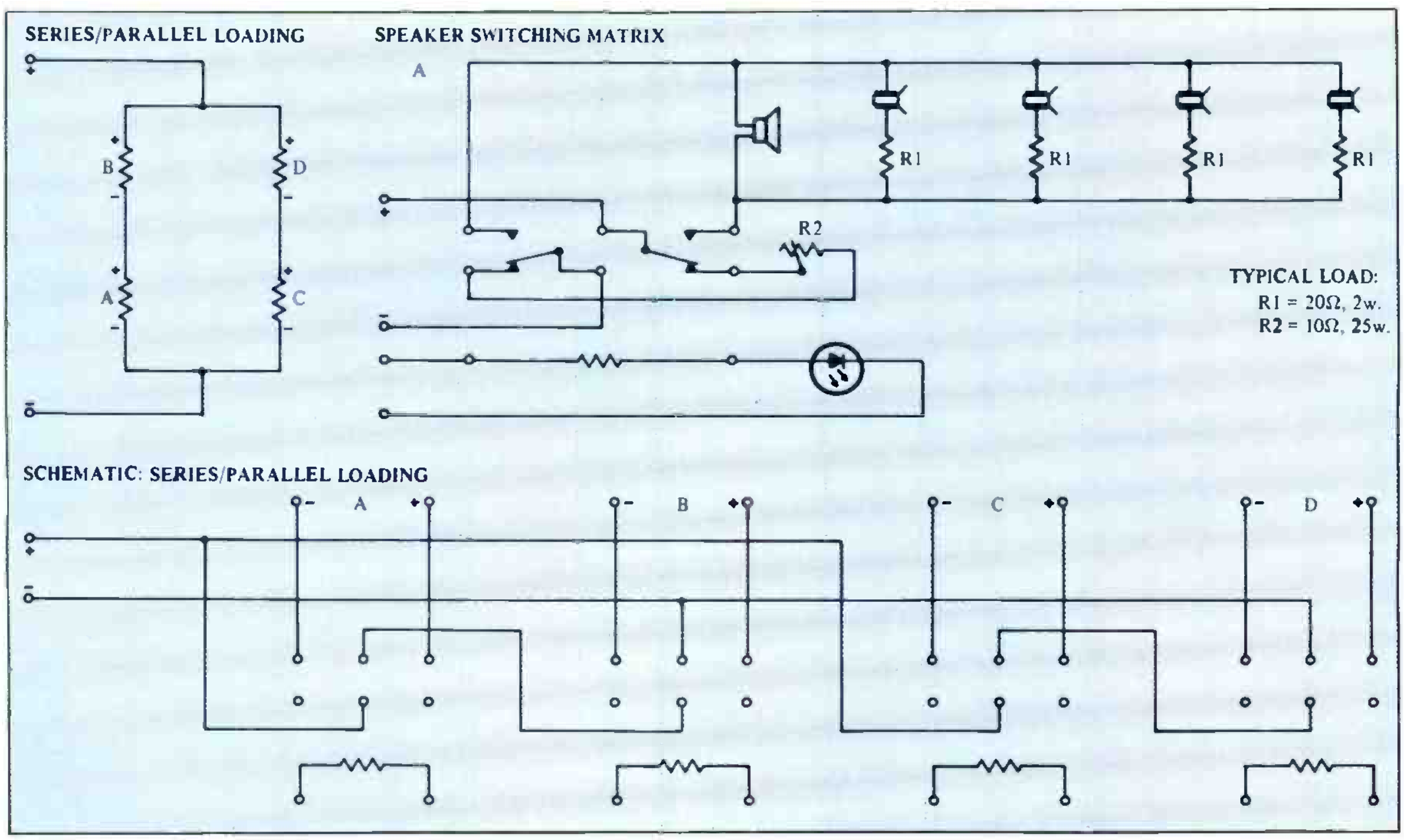


Figure 5. Speaker switching matrix used to develop several character point-sources from a single tape track.

JBL 4313s (pancakes swapped for PZTs) will also serve as reinforcement for the stage show, although we would find out later that the individual character systems would carry the room so well alone that very little reinforcement would ever be necessary.

The Stage Show System. Original specifications called for a ¼-inch eight-track open reel tape deck and speaker switching matrix that would route amplifier outputs to enclosure is not on a direct line-of-sight axis with the figure of event. The average pair of ears is very keen in identifying the localization information provided by a full range speaker system. This can result in cueing the eyes to, for instance, the barrel next to a figure, rather than the figure itself.

With A.I.A. techniques, this problem simply never occurs. We have found that when you separate the low and high frequency components of a full range system by

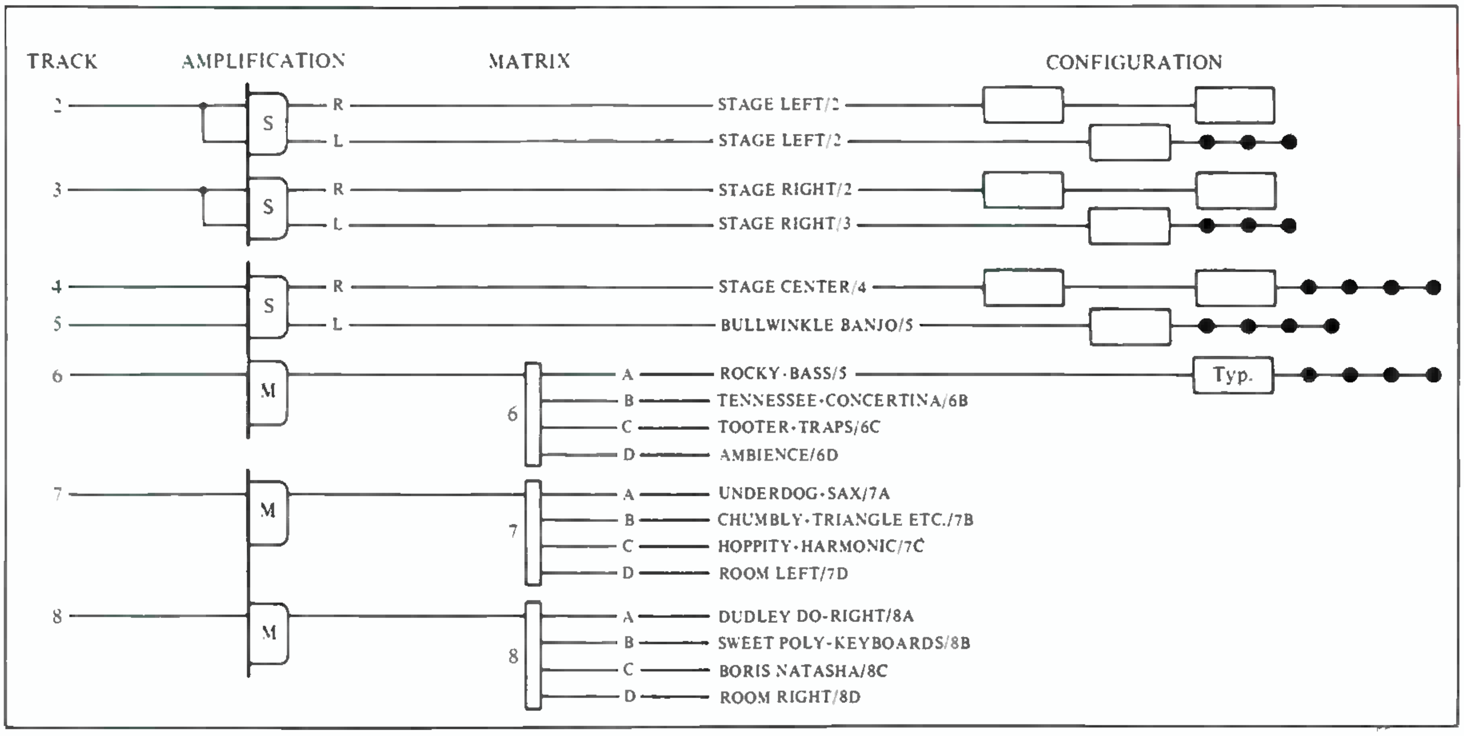


Figure 6. Bullwinkle's audio system block.

appropriate character systems as determined by MACS. This was quickly negotiated to one-inch eight-track and matrix, with later systems to be 16-track—hardwired to eliminate the matrix altogether.

The use of a relay switching matrix is a common belief of developing several character point-sources from a single tape track. (schematics appear in FIGURE 5).

Tape transports for the water and stage shows were Ampex frames "hot-rodded" especially for our prototype by Ron Newdoll and a dedicated crew at Accurate Sound.

The complete stage show audio system is diagrammed in FIGURE 6. Track one is address code for MACS; tracks two, three and four are stage left, right, and center, respectively; track five is Bullwinkle's (the star gets his own discrete system); tracks six, seven, and eight feed the matrices as shown.

PS Audio amps are used exclusively throughout the entire installation. Character systems, with the exception of Bullwinkle's, are matched JBL 4301Bs with A.I.A. assistance.

Bullwinkle's Character System, or "Antler Establishment." Bullwinkle's speaker system is bizarre, to say the least, even within the world of D.A. point-source techniques. And the A.I.A. application (see FIGURE 7) is only part of this unique configuration. For some reason, D.A. designs love to throw established audio engineering principles right out the window.

Normally, point-source enclosures would be installed facing toward the prospective listener. On occasion, this can actually distract the viewer if the point-source

several feet, and then set half of them in motion, the standard principles of time alignment, phase plane relationships, and even signal phasing, no longer apply. The A.I.A. array becomes the point-source, and we are free to apply static cone enclosures in almost any orientation relative to the figure. For our hero, two JBL 4313 enclosures were positioned facing backwards, aimed at the three-sided back walls of center stage forming a W or reflex bass system...of sorts. Because these two systems were applied symmetrically, they created a static center image that aligned surprisingly well, all by itself.

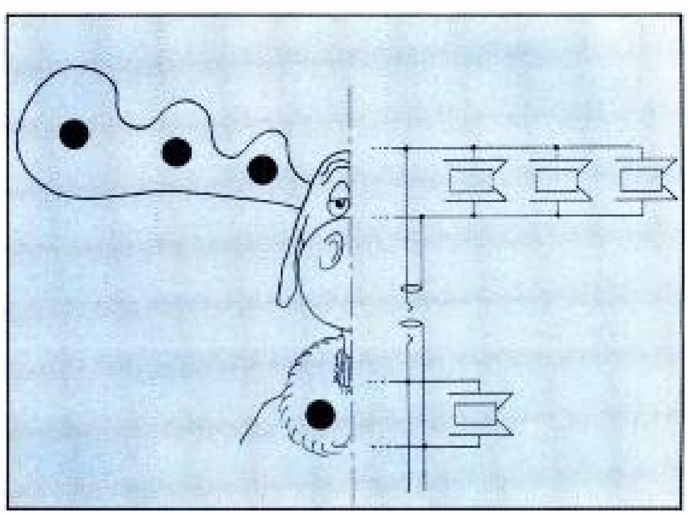


Figure 7. Bullwinkle's Aural Image Animation (A.I.A.) array.

The A.I.A. array creates an honest point-source image that changes in a natural and expected manner when Bullwinkle swings his torso from side to side, rocks back and forth on his barrel, or turns his head to look in any direction. This is the essence of A.I.A. mechanics. When properly applied, these subtle effects should go unnoticed; the subconscious mind simply accepts what it perceives to be correct. However, once noticed, or when pointed out as in our sonic demonstration, these same effects suddenly become quite apparent. By physically animating the speakers, we are recreating a spontaneous acoustic event in the room with the listener at every performance.

MOOSIC CREATION AND THE L.I.V.E. MIXDOWN

At some point, it became quite obvious that the unusual nature of the multi-point source/A.I.A. image would make mixdown in the studio difficult, if not impossible. the only way to fully anticipate the Calliope effect and assure its proper interpretation into the actual restaurant acoustics was to go there and use the show itself as a monitor.

With only the instrumental arrangements already recorded, all final pre-production was accomplished in the last few weeks before the show installation. The entire production team became quite animated in preparation for the all-too-rapidly approaching three day window between show installation and programming. All music for the Bullwinkle Restaurant Showcase was composed, arranged, conducted, and produced by a talented film and TV scoring producer, William Broughton, who was busy orchestrating final arrangements for voice talent, dialogue recording sessions, and dialogue/sound FX dubbing. Scott Hennesey was busy developing an eleventh-hour script that literally pulled it all together, although how he ever did it is a moostery to me!

We were extremely fortunate to have Bill Scott (the voices of Bullwinkle and Dudley Do-Right, and co-producer with Jay Ward of the original T.V. cartoon show) and June Forray (the voices of Rocky, Natasha, and countless others) recreate their original characters for our stage show. We were also fortunate, in the absence of other original voices, to have Dave Stamey (Tennessee Tuxedo), John Swanson (Boris Badinov), and Corey Burton, who recreated the late Hans Conried's Snidely Whiplash with unnerving accuracy.

With music and voice tracks in the can, Bill Broughton, Sound FX editor Steve Hope, and Ed Barton (engineer) set about the business of dubbing all the various pieces down into 24-track, two-inch format, while final preparations were made to turn the restaurant into a recording studio.

MIXDOWN

Day One. The recording equipment arrived from San Francisco at just after 6:00 AM, and was in place and operational before 8:00 AM. It certainly was the right stuff (FIGURE 8). Our L.I.V.E. studio consisted of an Ampex MM-1100 24-track, two-inch machine, the Harrison "Alive" console in the 32 × 8 × 4 configuration, an Otari eight-track, one-inch transport for mastering, and slightly over seven feet of 19-inch rack space filled with signal processing equipment. The outputs of the seven audio channels were connected, via an umbilical, directly to the show amplifier inputs.

L.I.V.E. "enginears" Daniel Prothero, Dan Scharf, and Ed Ryba spent the rest of the day aligning the entire audio chain to the show systems, re-calibrating every channel of both decks, and beginning the process of building voice patches for each character system. The exact nature of this procedure is proprietary and must be kept a trade secret, but the advantage of mastering your product on the audio system to be used to recreate it in the field should be obvious.

Figure 8. The L.I.V.E. mixdown studio "enginears" Daniel Prothero and Ed Ryba build character voices with the assistance of MACS.

Direct interpretation eliminates guesswork; all EQ and other processing is a direct function of the mixdown. Because this signal processing is already encoded onto each track of the master, there is no need to provide this equipment for each show in the field (the expense would be substantial).

Day Two. Production call was 8:00 AM on the second day. Daniel and Ed finished the last minute tweeking, and we were ready to begin making mixdown passes for the seven show segments chosen for the grand opening. Admittedly, the L.I.V.E. mixdown is an elaborate process, and it seems to grow more sophisticated with each application. However, direct interpretation is simply listening, and once the voice patches are in place, the engineering approach is similar to that of a live concert. By 8:00 PM the night of day two, we were finished with the mixdown passes, had edited the show segments together in sequence, and had provided Fred Hope and Doug Mobley (and MACS) with a work print ready for addressing and programming the moves, the water show, the lights, the curtains....

Day Three. With our studio back in road cases, there was nothing else to do but have a pizza, sit back, and enjoy the show.

SUMMARY

It is the goal of the D.A. designer to tell a story. In general, we must accomplish this by catering to just two senses: the eyes and the ears. The sheer complexity of creating the visual aspect of a dimensional animation often overshadows the importance of the aural aspect, and yet it is equally capable of telling a story.

If I have told my story well, I have shed some light on a little known and little understood medium. The next time you're riding through a theme park attraction or dark ride, maybe you'll notice that the sound of an animated figure is coming from the box next to that figure. If you don't...then maybe I did tell my story well. Happy listening. ■

GEORGE STEPHENS

Modular Sound Meets Its Waterloo (Folk Festival)

Modular Sound Reinforcement is proof positive that "more" is not necessarily better—or louder!

MAKING A PROFIT in the concert industry often requires finding a production company that can do the best job of combining high quality with effective cost saving factors. The Waterloo Folk Festival that took place Labor Day weekend 1983 provided a good opportunity to see (and hear) a quality system with an innovative approach in action.

Waterloo Village, in Stanhope, New Jersey, is an authentically restored Early American town in a pastoral setting—a scene totally unfamiliar to travelers of the New Jersey Turnpike. For the last few years, Doug Smith has directed a popular concert series that is held in a large circus tent on one side of the village. The 1983 concert series featured more shows, bigger names, and larger audiences than in past years. Highlighting the summer was the Waterloo Folk Festival—a two day extravaganza featuring Judy Collins, Don McLean, Jorma Kaukonen, Melanie, Roger McGuinn, Tom Chapin, Richie Havens, and many others, all on one stage. During the second day local artists would be featured on another stage at a different location.

The concert site posed a bit of an acoustical problem. The tent was large enough to house a three ring circus—approximately 150 feet wide and 100 feet deep, with three massive poles down the center. The 60-by 40-ft. stage was permanently installed along one side with a concrete floor sloped from all points in the tent down to the stage. Up to 2500 folding chairs could be placed on the floor, while another 5,000 could be accommodated in the field surrounding the tent. This situation meant that as much sound as possible had to be directed outside the tent. However, too much sound directed into the ceiling resulted in a heavy slap in the 300 Hz-1 kHz region at the back of the tent.

To handle those sound reinforcement requirements, Doug chose Modular Sound Reinforcement of Princeton Junction, New Jersey.

According to Modular's crew chief Robyn Gately, "Many people approach [this type of situation by] bringing in tons of equipment, believing that the peaks and valleys will eventually even out. Unfortunately, this effect is like pouring a dump truck of pebbles into a pond; you may have waves everywhere, but you can't pick any of them out individually. We are extremely aware that the key to overcoming reverb is to have the most efficient, aligned cabinets available. The fewer the cabinets and the more cohesive the wave front, the less clutter there is. As a result, we may not be the sound company for people who expect mountainous stacks. We've occasionally lost shows because people didn't believe our equipment was physically big enough to do the job."

The speaker system, sidefill monitors, and all amps for the house PA at Waterloo Village were placed on one 5-ft. high section of scaffolding on each side of the stage.

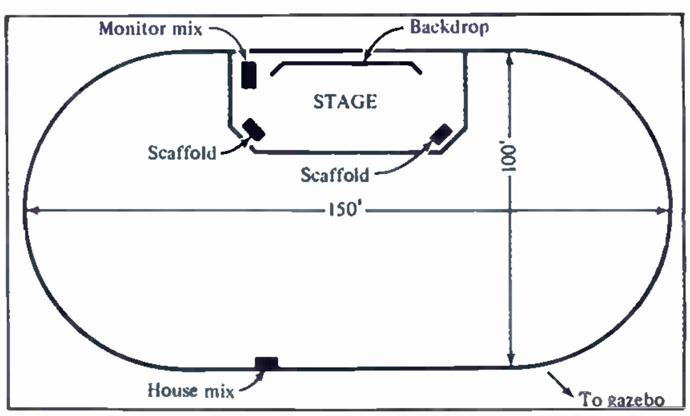
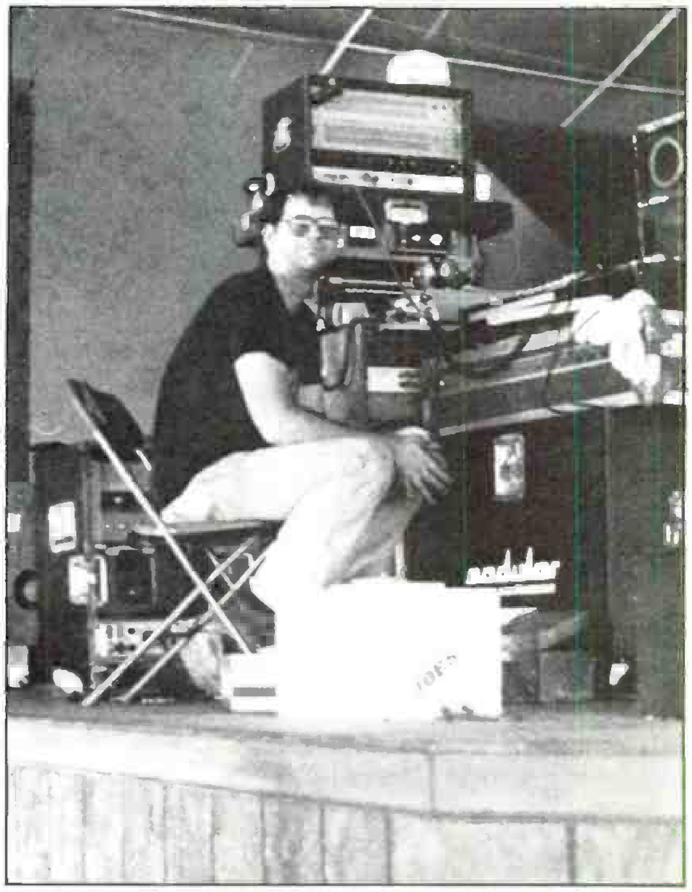


Figure 1. House mix position diagram.

George Stephens is a free-lance audio writer based in the NYC area.



Monitor engineer Vann Weller at the 2005 A.D. console.

"Although hanging the cabinets would have been preferable, there is no easy way to do it," said Vann Weller, the monitor engineer. "The tent poles sway tremendously in the wind. Being in a storm in here is quite an experience. We're considering using Genie towers next year, but the scaffolding is a quick, simple solution that allows the sale of several hundred more seats than would have been possible with a bulkier system placed on the stage."

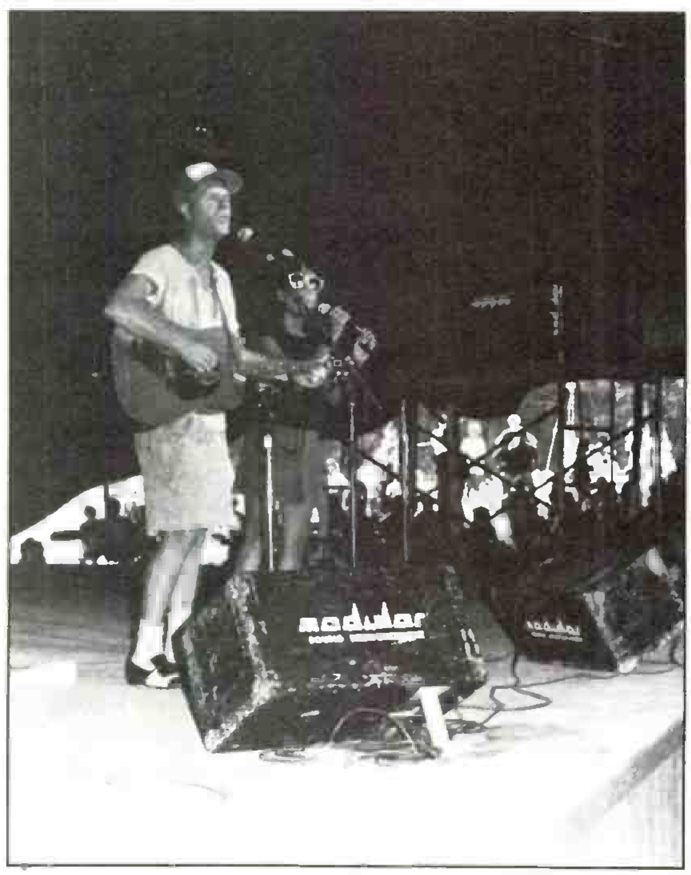
THE SPEAKER SYSTEM

Modular's speaker system for the festival consisted of a new two-box four-way design by Joe's Sound & Salami Co. of Trenton, NJ. Modular is one division of Joe's Amalgamated Industries, Inc., while the cabinet manufacturing company is another. Modular's President George Williamson says, "Our concept is that a speaker system should be like a clear window. The working relationship with a cabinet builder allows us to specify the parameters of a box and know they are going to give us what we expected. As a result, when we were choosing a new system, the quality of sound was the first criterion that had to be met. The new boxes had to be aligned; at the same time field replaceability of drivers was, as always, a concern. The drivers had to be made by a major manufacturer, because we wanted to be able to go anywhere in the world without the worry of recones or replacements.

"The other essential factor was space efficiency, due to transport cost. Many people have proven that Bass Reflex cabinets offer the greatest output to space ratios, so we wanted the flattest, most efficient reflex cabinets possible. One line manufactured by Joe's uses an exclusive 'wood-resin alloy' that creates a tighter, more efficient cabinet than standard birch boxes. These are the 'Second Generation' of aligned PA cabinets. The cabinets use the Joe's *SLIDE-ALIGN* (patent pending) technique to move the speaker components into perfect alignment at the show. The sound of these boxes and the fact that they were designed for JBL drivers means they meet all of our criteria. In fact, a typical system for 3,000 seats with on-stage monitors and all electronics can easily fit in a cube van."

The speaker system used at the Folk Festival was comprised of a subwoofer system (two boxes per side) each containing two JBL K-151 or 2240 18-in. drivers, and a three-way high-end box (two per side) that picked up at the 250 Hz subwoofer crossover point. The second box, called an "Atomic Pile," is comprised of two JBL K-120 or 2202 12-in. drivers (250 Hz-1.2 kHz), six JBL 2105 5-in. drivers (1.2-5.2 kHz), and two JBL 2425 compression drivers on JBL 2307/2308 horn/lenses for super-highs. "The use of the 5-in. drivers for this kind of show helps a lot. Some people find the forced-air sound of midrange horns very objectionable," states Gately. "Some of the acts are very specific about what they will use. The five-inch drivers do need about half a millisecond of delay to bring them into perfect crossover alignment, but they are the only Joe's box that needs any electronic correction." When asked why the three-way box was called an "Atomic Pile," Gately replied, "The subwoofers were originally named 'Multiple Warheads' because of their devastating effect. When we built the new box we figured that if you put an 'Atomic Pile' on top of a 'Multiple Warhead," you had a killer PA!"

All of the House amplifers for the Festival were Phase Linear Dual 500s. Developing greater than 1200 watts-per-channel into 2 ohms, four amplifiers drove the entire system. These amps were chosen due to Modular's success with Phase Linear 700 amps. Realizing that distortion kills drivers, the company felt the less clipping seen by a

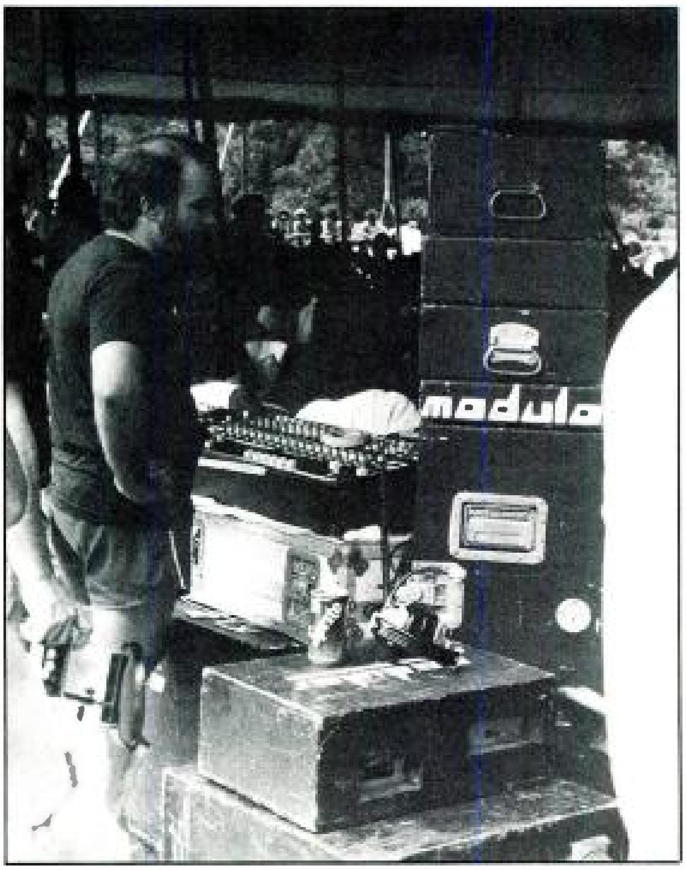


Loudon Wainwright III with Suzzy Roche on the main stage. (All photos by Beverly Morgan.)

speaker, the better. With over 10,000 watts available, clipping was not a problem. All amps were cased with front panel patching manufactured by Joe's.

HOUSE MIX

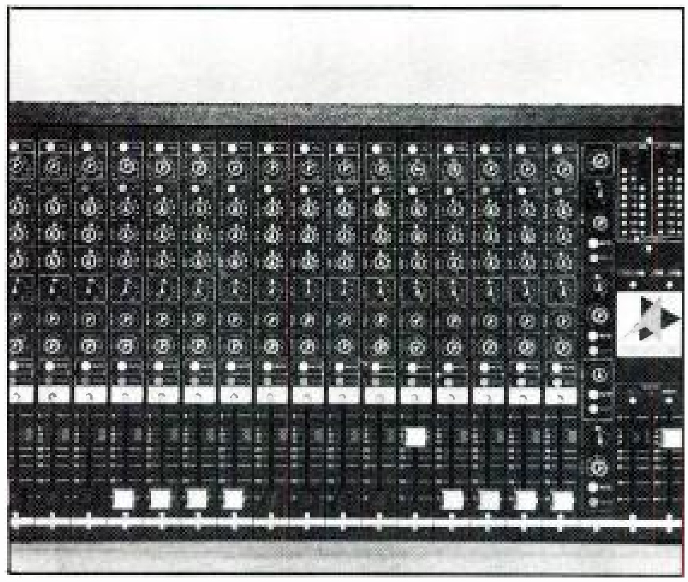
The House mix position (see FIGURE 1) offered another interesting approach to system design. The company can obtain an type of board desired, but the house engineer's personal preference is for the 2005 A.D. console. "The board is simply designed, but effective," reports George Williamson. "It doesn't have submasters or complex equalizers on each input; rather, the designers felt that it was smarter to let extensive EQ or limiting be patched into each input pre or post EQ, or across the mix output summing amp. This simplicity results in some unbeievable phase and noise specs. The board has been used on many gold albums, including projects by Jackson Browne and Bowie." Originally manufactured in Philadelphia by John Buffington and Lance Strickler, warranty and production for the 2005 A.D. is handled by Joe's.



House engineer Robyn Gately.

Modular's philosophy of "simplicity works best" means that they try to keep nothing between the board, crossover, and amps. The company tries to eliminate the repetition of needless Input/Output amps wherever possible. Since patching is done at the summing amp, the dbx model 118 limiter was chosen. The engineers found it to be a cleaner unit than the dbx 160 in this particular application. Filling out the equipment at the house position was a Klark-Teknik DN27 equalizer, two DeltaLab DL-4 delays, an AKG BX-5 reverb, several more dbx 118 limiters, and two Technics cassette decks. Interestingly enough, only two engineers at the Festival even switched in the EQ—quite a statement about the quality of the system.

The house mix position was placed on the far side of the tent from the stage, slightly off center and under the


The 2005 A.D. console.

overhang of the canvas roof behind the last row of seats. "This position was chosen because it was out of the way and provided us with a good representation of how the acoustics were affecting the sound," said Gately. "Because it was one of the worst positions in the house, we knew everywhere else had to sound better. It also gave us the chance to show how an aligned system reduced reverb clutter compared to a non-aligned one."

Monitors for the show were handled by Vann Weller, utilizing another 2005 A.D. console modified by monitors. The most complex requirements were for Judy Collins, so the system was set up according to the specifications of her soundman, Dick Zicari, and then modified for the other acts. Mix #1, the sidefills, consisted of two 15-in. JBL 2220s in Joe's "Air Activator" cabinets for lows, with a JBL 2441 on a 2395 Horn/Lense for highs on each side. These units were placed up out of the way on the scaffolding. All other monitors used for the Festival were Joe's "Stage Eliminators," an aligned box using JBL 12-in. drivers and JBL 2425 or 2470 high drivers on 2301 Horn/lenses. Mix #2 was assigned to the front vocal monitors, Mix #3 to the drums or piano, and Mix #4 assigned to the piano or secondary vocals. All monitors were powered by Phase Linear 700 amplifiers, with the exception of the sidefill highs, which used a Crown 150. One-third octave EQs from UREI were available for every mix.

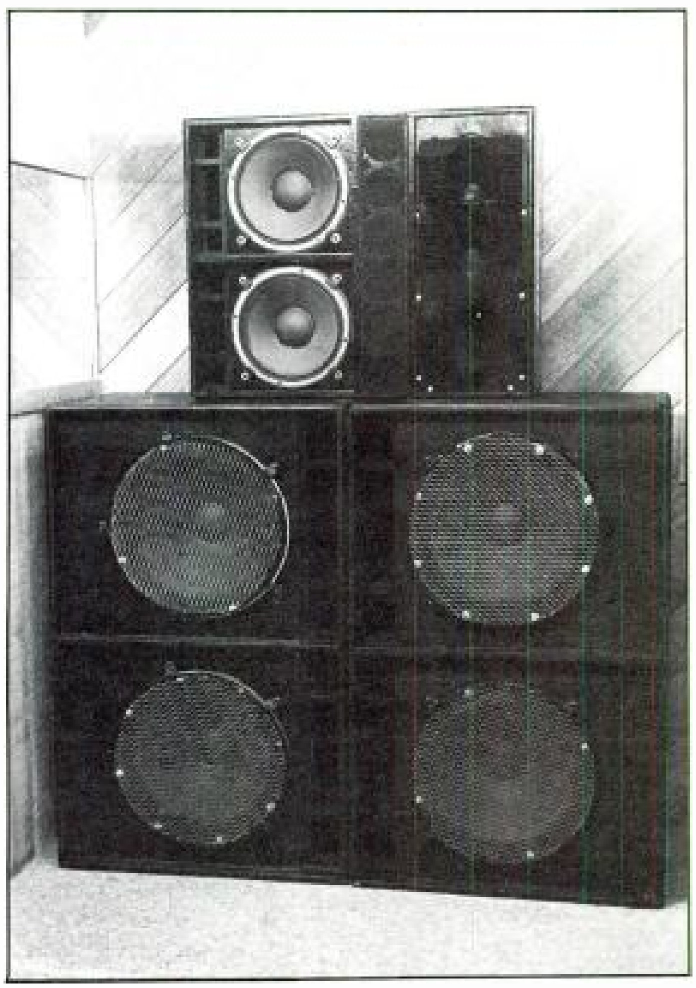
"All tone control for each instrument could be achieved by the three-band EQ on the board inputs," said Weller. "Because of the flatness of the aligned monitors, the graphic EQs were not needed except in very high-gain vocal mic situations with the weakest singers, or when the acoustic guitar mics were positioned low and angled downward. We tried to let all the peformers do whatever was most comfortable for them, and then work around that. As it turned out, most of the mixes never used the EQ throughout the whole festival." Additional monitor equipment included plenty of dbx 118 limiters, which were never patched in. A Talkback mic led from the house console into the monitor system so communications could be established even if no one was near the Clear-com intercom.

ON-STAGE

When stage cabling was laid everything was based around Judy Collins' stage set-up. This meant everything had to be planned around the fact that no cables were allowed downstage during her set. The decision was to place the snake behind the backdrop, with sub-snakes carrying the signal where needed. Since pre-planning for all of the acts was an impossibility, it was decided to standardize mic'ing wherever possible at the pre-production meeting. Alan Reinhardt, the stage technician, was assigned to meet with the acts immediately after their arrival to get their stage requirements. Because of the large number of acts and the length of their sets, promoter Doug Smith wanted to keep the pace up by having less than five minutes in between acts. Through careful planning this time table was adhered to, except for the occasional artist delay.

Visiting soundmen found they could have just about anything they wanted in microphones. After hearing the system, most went with what was already in use: Vocals—Beyer M-400s or Shure SM-57s; Guitars—Beyer M-400s or Sennheisher 421s or 441s; Piano—Sony ECM-150s and Sennheiser 441s; Bass and Keyboards—DI; Drums—Sennheiser 421s for all Toms, AKG D-12 for Kick, Sennheiser 441 for Snare, and various condensers for all cymbals and percussion. For the set by Judy Collins, soundman Dick Zicari brought in AKG 451s. The basic premise is that all EQ starts where the sound is converted to electricity, so it is easier to get the desired result if the engineer chooses his mics carefully. Most soundmen were amazed by the sound achieved through careful placement of the Sony omni mic on the piano, as the piano was loud and clear with no feedback.

During the second day of the festival, a program of local folk artists was held at the Gazebo near the middle of the village, about half a mile from the main stage. Knowing that the audience for this event would number no more than 500, Modular provided a bi-amplified PA consisting of a JBL 15-in. 2220 in a Joe's cabinet for lows on each side, with a JBL 2441 for highs. Amplifiers were again Phase Linear 700s, while the console was another 2005 A.D. Only one monitor send was necessary, and the two Stage Eliminators provided more than enough monitor gain, according to the musicians who used the system.



One side of a typical speaker system for rock'n'roll—2,500 seats.

FAST SET-UP, FAST GETAWAY

System logistics provided a real challenge for the staff of Modular. Being a local sound company that does all kinds of shows from the smallest to the largest (the company has done venues up to 57,000 seats), the management tries to keep the equipment out on the road working as much as possible. For this particular show the monitor system was set up the day before the show, the Main stacks came from a 2,000 seat disco the morning of the first show, and the Gazebo system arrived on Sunday after a concert in Southern New Jersey the night before. The only problem came when the disco ran two hours longer than expected. Even though the Main stacks did not arrive at the concert site until almost 8 A.M., first sound check took place at 9:30 as scheduled!

At the end of the Festival, some of the artists were surprised to see the sound truck leaving before they did. This was especially interesting considering there was no loading crew, just the four sound company employees assigned to the show. As president Williamson explains, "Almost all of our load-outs are completed within an hour after the show ends. Part of this quickness is a continuation of the 'simple, basic' concept applied to packaging, and part is due to the experience of doing dozens of shows per week. Of course, it helps that we have our R & D department see as many shows as possible. In addition, we try to have a class for all of our engineers each week. This helps us standardize our crews and promotes a constant exchange of knowledge. We constantly emphasize the importance of accommodating the artist and maintaining clear sightlines."

WRAP-UP

The engineers at Modular first became interested in improving the state of the art of PA after hearing the Grateful Dead's "Wall of Sound" many years ago. Although they were very impressed by the quality of the sound, they were not too thrilled with the trucking costs. According to Robyn Gately, "We're at the point where we feel we've accomplished many of the things we set out to do. However, there are still significant advances that can be made to improve quality and increase profits."

Discussing the use of the system he added, "We find it's easier to mix on our type of system if the engineer is able to look beyond the 'mystique' of conventional, mountainous PAs to what the future holds. Studio experience definitely helps in adjusting. People used to high distortion levels turn it up absurdly loud unless they have a sound level meter next to them."

The Waterloo Folk Festival '83 was successful in part because of the experience and professionalism of the sound company. Planning played a large part in this. However, after the first day's show, the lighting crew was surprised to see Modular had a portable VCR, Colecovision, and a television. It was obvious that this was not the typical jaded sound crew; these engineers still knew how to have fun. ■

JESSE KLAPHOLZ

On The Boardwalk

Roll the dice and take a walk along Atlantic City's famed Boardwalk in search of—what else?—sound reinforcement systems.

THE ROLE OF an audio engineer in the Atlantic City casinos is much more involved than you'd expect. Our story begins by placing into perspective just where entertainment fits into the casino scene. Although the entertainment departments' policies differ from one casino to another, the role of the audio engineer is pretty much consistent.

It's 8:00 P.M., and we're driving into Atlantic City. The skyline would make a perfect opening scene for a movie. (Unfortunately, Paramount stole my idea.) After a few minutes of looking around, we find ourselves on that road paved with gold: Pacific Avenue! (That should bring back some memories for those who were into Monopoly.) In case anyone hasn't been paying attention to the news, Atlantic City has been steadily evolving into the Entertainment Capital of America and perhaps the Gambling Capital of the World. People flock to the casinos any way they can; last year 340,000 charter buses brought more than 11 million gamblers to the 12,209 slot machines and over 1,000 tables and other games. In 1983, gamblers at the nine Atlantic City casinos lost more than $1.7 billion!

Unlike Nevada's laws permitting twenty-four hour gambling and slot machines everywhere, the gambling here is restricted to the casinos, from 10:00 A.M. until 4:00 A.M. on weekdays, and until 6:00 A.M. on weekends and holidays.

The first adventure of the evening is a game called Parking. After buying Parking, you may proceed to any one of the casino floors.

CASINO NOISE?

The casino floor is where all the action is—the battleground where the game takes place. We've all heard of pink noise and white noise, but here there is something different: casino noise (1). Casino noise is very complex in nature: it's primarily comprised of a base of pink noise with many impulse noise sources superimposed.

The Yamaha PM-2000 24 x 8 console in use at Billy's Pub in Bally's Park Place Casino Hotel.

The superimposed noise sources found in casino noise fall into two main categories (2): human-generated sound and mechanically generated sound. The human-generated sounds can be extremely varied in both frequency range and sound intensity level, depending on who just won the jackpot at the slots! The mechanically generated sounds are mostly dominated by a sound similar to the repetitive sound of a wooden clapper, such as the ones used in reverb analysis of rooms; its source is The Big Six Wheel.

The most intriguing aspect of casino noise is its property of drawing you into the action. Casino noise creates electricity in the air (no, not the kind Ben Franklin found with his kite) that puts you right into James Bond's world.

BOARDWALK AND PARK PLACE

How often during a session of Monopoly did we battle for those properties! Now, Bally's Park Place majestically stands on this historically sought after property, with over 60,000 square feet of casino floor.

Jesse Klapholz runs an audio consulting firm specializing in acoustical analysis and design in the Philadelphia area.

1. Duck, Don L. "Noise in Gaming Establishments." *Journal of Noise*, Vol. 1 #1.
2. Hosehead, Dr. P. Tech Note #408, "The Nature of Casino Noise," *CompCom Newsletter #34*, 1983.

We start our expedition here at one of Bally's lounges. We're taking our time getting to the lounge, because while we're on the casino floor, drinks are free! There's a sound system here—elementary, but it sounds good, and everybody seems to be happy. The waitress comes over, so we ask her if there is any other entertainment around. She laughs, "Are you kidding? This is Atlantic City—we have anything you want! Just check out Billy's Pub next to the Deli." (The *deli*?)

THE HARDWARE

As we walk into the club we can't help but notice the spacious light and sound booth. A Yamaha PM-2000 24×8 with both a light and a sound man! Outrageous!

The stage is a circular thrust-type design four feet off the floor, and is dramatically backlit. The club has all the stage lighting and special effects you'd expect in a conventional theatre, all run by a manual two-scene board. There are two bars on either side of the room with spacious and posh seating in the center. The stage has a very clean and slick look, unlike many of these types of lounges.

We watched the first group, a five piece fusion-jazz-rock band, and sat through the first set change to a fifties rock'n'roll show band. The set change clocked in under fifteen minutes. But things didn't start out like this in the beginning.

The head sound technician of Billy's Pub, Gary Battestella, reminisces: "When I first came here, there was a small eight-channel mixer right on stage, your typical lounge system, and there were two small 'stage wash' monitors. What you see here now is the result of requesting changes through all the proper channels, getting budget approvals, and so on; it takes a lot of effort, planning, explaining, and patience."

Gary explains how and why the system evolved: "When the lounge was originally built, the intended use was as a piano bar-type lounge with trios and light vocal acts. But as time went on the entertainment policy started changing over to more high-energy rock and show group-type acts. With the entertainment going to a more high-energy context, the system(s) had to change. First, the booth had to be built with all the mic lines run and a bigger board installed, a PM-1000 16×4. The original house sound system used 8 Altec 604s driven by an Altec incremental amp (that's 75 watts per cabinet). We beefed that up by installing two BGW 750s for the lows and two BGW 250s for the highs and used a BGW electronic crossover. Later we got the four dbx 160X limiters, a Master Room reverb, a DeltaLab digital delay, and all Klark-Teknik third/octave equalizers."

The technicians were thrilled when they found out that through an equipment shuffle they were to inherit a PM-2000 24×8. The acquisition of the 2000 rounded out the system, enabling the sound crew to give the performers four monitor mixes, and change from one act to another more efficiently.

The stage in Billy's Pub is set up like a recording studio. There is a house set of drums judiciously mic'ed up behind a plexiglass baffle, a Yamaha CP80, a Fender Rhodes, and an EV S15-3 speaker system. To the left of the drum-booth is a house bass system: a fifteen-inch JBL driven by a BGW 500 and controlled by an Ashley parametric EQ preamp. There are mic input panels all around the stage which have both XLR mic connectors and parallel phone jack inputs (transformer balanced).

All the stage instruments are run direct when possible, and then routed through the four foldbacks of the 2000 to the respective monitors. The keys, drummer, and guitar players each have their own mix, leaving one mix for the front vocals. All the monitors are custom built by the sound crew using EV, JBL, and Community Light & Sound components, and are mounted in the ceiling above the stage to keep things neat.

CIGARS, CIGARETTES...

After three sets at Billy's Pub, we decide to continue our adventure. Strolling on the Boardwalk, breathing the invigorating salt air, we find our way into another casino. We get the bug and stop to play the slots. "Susie Salem, please go to Craps Pit #9," we suddenly hear. Wow, that paging system really cuts through all the noise. That's pretty hip!

All casinos have fairly complex paging systems that must meet the requirements of the Casino Control Commission. Besides functioning as twenty-four hour background music and "in house" advertising, the system must be capable of being heard and understood anywhere in the casino-hotel complex in case of an emergency situation. In fact, any time a fire alarm is activated, music inputs to all systems in the lounges and theatres, as well as background music and paging system are automatically muted.

A quick tour of any of the equipment rooms running these systems will reveal many more racks of state-of-the-art equipment than you'd expect. You won't find the typical 30-watt, 70-volt mixer amp hooked up to a slew of cheap speakers. What you will find are mixers, limiters, noise level sensing and adjusting systems, equalizers, banks of large, heavy duty amplifiers, and patch bays, all broken down into numerous area "zones." Similarly, a quick inspection of the ceiling will reveal high-quality eight-inch coaxial loudspeakers—and plenty of them. In fact, one casino is using Klipsch Heresy loudspeakers (a 12-in. with mid-range horn and compression tweeter system).

"Susie Salem, please go to Black Jack pit #7," comes over the paging system. Now who is this Susie Salem they've been paging all night? To satisfy our curiosity we asked one of the bartenders. She replied, "Why, she works with Karen Kent, Wanda Winston, and Marilyn Marlboro—they're the cigarette girls!" Well *excuuuse* me!

I'LL BE THE RACECAR...NO, THE HAT

We head back to Boardwalk, roll the dice, and end up at North Carolina Avenue, home of Resorts International Hotel & Casino, where the birth of Atlantic City gambling took place on Memorial Day Weekend, 1978. Naturally, entertainment's pace was set here, with a heavy commitment to quality big name acts. Frank Sinatra, Diana Ross, Sheena Easton, Stevie Wonder, Kool and the Gang, The Beach Boys, Pavarotti, and the N.Y. Philharmonic Orchestra have appeared in The Superstar Theatre.

Prior to the hotel becoming a casino, it was The Haddon Hall Hotel; the Pennsylvania Room (a ballroom) is where the Superstar Theatre now resides. The stage was originally much smaller, the seating was on a flat floor, and there was only an overhead distributed sound system. Since then, the stage has been enlarged to accommodate any event, from a full Vegas-type stage show/revue, to ballet, to boxing and video shots. The floor was built up to multi-levels, and the overhead distributed system was, of course, replaced with a concert-type sound system. The layout of the stage offers flexibility and efficient operation; it has lots of on-stage storage and a truck-sized freight elevator. There is a band cart at the rear center of the stage that can roll right onto the stage with a full band setup on it, or the band can play behind a "scrim" curtain when necessary.

A-1 Audio was responsible for the original installation, which consisted of a central cluster of four JBL 4350s and two side fill clusters of two JBL 4350s each. The loudspeakers were bi-amped and crossed over at 250 Hz, with a BGW 750 driving each 4350. There is also a rear-delay fill system consisting of JBL 4301s.

Clint Pease, the head sound technician of The Superstar Theatre, talks about the changes that he's made subsequent to the original installation: "The PM-1000 was replaced with a larger PM-2000 to accommodate more inputs and the added flexibility, and a Yamaha M512 was added as a submixer, as well as a PM-180. We've installed a sub-woofer system with four 15-inch JBLs and UREI EQ and signal processing to beef up the low end in the room.

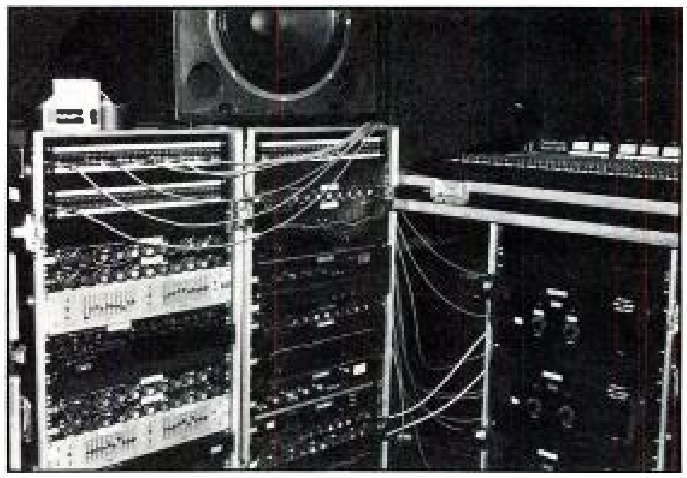
Walking into the booth at Resorts, you can't help but be amazed at the complexity and variety of equipment there. According to Clint, "This room is a huge night club, and it's fairly dead acoustically—making it ideal for rock and R&B-type acts. But when we need to liven the room up for someone like Pavarotti or Roberta Peters, the AKG BX-10, Lexicon 224X, and Prime Time 93 earn their keep. Recently we got this Aphex rack system with several of the CX-1 limiter/expander modules, and it's the best limiter I've used so far. The CX-1's combined limiter and gating functions can make a vocalist with poor mic technique sound like a pro, really smooth and natural. Using the CX-1 on drums is great; the limiting and gating functions in this application control the 'ring' and 'tighten up' the sound."

On some of the bigger shows, it is not unusual to use up fifty inputs on the system. Clint explains, "When you individually mic a large string section, a full horn section, a rhythm section, a multi-keyboard setup and the vocals, that eats the inputs up really fast."

Discussing his technique, Clint says, "I use two stereo subgroups and a stereo return (for effects). I mono the vocals and the lower frequency instruments, and then control the imaging of the other instruments on stage. I had been using two Countrymen EM-101s and a Helpenstill pickup in the piano. But the other night when Enzo Stuarti was in, I tried our new C-Tape in stereo. It not only eliminated some of the problems with my older setup, but also improved the sound a lot."

We asked Clint what the most challenging show was that he's done at Resorts. He replied, "We did a one hour long live cable T.V. show here for fourteen weeks, called Atlantic City Alive. Each show featured five to six acts, anything from an animal act to a singer with a full orchestra. Everything had to be completely mic'ed up including audience mics, with a split to the video truck. We had an extensive communications setup to be concerned with also. The setup and rehearsal started early in the morning and we worked straight through the 7:00 P.M. air time, and then after the show, broke the whole thing down!"

What is the key ingredient to being an audio engineer in the casinos? Clint had this to say, "You can't lose sight of what your job really is, and that's to make sure that when Don Rickles or Johnny Carson or Frank Sinatra goes out on that stage, that mic better work, and everybody in the room better hear and understand that performer. In every decision made concerning equipment purchases, system changes, and setup in general, always keep reliability in mind."



Some of the hardware in use at the Royal Swan Ballroom in the Tropicana Hotel.

MOVING RIGHT ALONG

We take the Boardwalk Tram, one of Atlantic City's attractions, and the name Tropicana in flashing colors catches our eyes. The Tropicana is the newest and most elegant casino in Atlantic City, and is the world's most expensive hotel/casino , at a total initial investment of $340 million. Currently under construction is a 1,700-seat theatre, scheduled for completion in January, 1985, at a cost of $10 million. In the meantime, the Entertainment Department has set up camp in the Royal Swan Ballroom.

The ballroom is 96-ft. wide × 200-ft. long, with an 18-ft. ceiling and an RT60 of more than three seconds in the midrange. The room is outfitted with the latest state-of-the-art equipment, and has mic inputs, video lines, and multiple communications systems throughout the complex. The overhead distributed sound system consists of Altec 12-inch coaxial loudspeakers driven by BGW amplifiers; still, it was decided upon by the Entertainment Department to augment this with another sound system.

TekCom Corporation of Philadelphia supplied The Tropicana, with a three-way CL&S split proscenium system and two two-way delay stacks. This system, along with monitors, signal processors, and several wireless microphones, did the trick for the opening revue, Monte Carlo. After six months of Monte Carlo, The Trop brought in the Broadway production, Can Can, which required twelve wireless body-pacs, a diversity antenna system, and more signal processing.

A year later, the entertainment crew was given two month's notice that a star policy was going into effect. The audio department had to make sure that they could meet all the requirements of the incoming groups. Patrick Baltzell got together with TekCom again and put together a good system to fill a tall bill.

Pat talks about the requirements and what they did; "Since the room is a ballroom, it is used for conventions and boxing events all week long. When a star comes in here for a weekend, we have to be able to set up the system in four hours, and then be ready for a sound check and rehearsal. The most important criterion, therefore, was that the system be entirely portable. We started with our custom-made Wireworks fifty-pair snake, built around a central box which has nine sub-snakes that plug into it. Then a 250-ft. fifty-pair comes out to the house racks and a 75-ft. fifty-pair split goes to the monitor racks. All the fifty-pair connections are made with the large zero-insertion force-type AMP connectors. The house racks have all the mic inputs plugged into a patch bay, and then fed over to the two Yamaha 1516 consoles. The monitor system consists of a Soundcraft 400B 24×8 board and UREI, Audioarts, and Altec signal processing. We have two sets of sidefill monitors; the downstage set is a three-way CL&S system with an 18-in. JBL, a 12-in. JBL mid-bass horn, and a JBL 2445 on a PC-494 constant coverage horn, all physically time-aligned with FFT analysis. The two upstage monitors are standard two-way loudspeakers for the orchestra. We also have some interesting front spots made with a 12-in. JBL and a JBL 2445 on a CL&S PC-494 horn, bi-amped and time offset-corrected with the Professional Audio Systems (PAS) TOC-23 electronic crossover. With all these improvements, we have been able to satisfy all the requirements of the groups' riders."

Pat's job is a little different from the audio techs at other casinos in that he is responsible for the smooth operation of all the audio systems: background music and paging (including making recorded announcements), lounge sound, convention systems, special events, and, of course, sound for the stars. One such special event was this past summer's July 4th celebration on the beach. Pat was responsible for supervising the setup, and mixing Peter Nero and the Philly Pops Orchestra, Leslie Gore, and Chubby Checker.

With the Tropicana theatre now under construction, Pat has been involved with the project from its start. For the entertainment staff and executives, he recently set up a demonstration of the top manufacturers' components and systems for both FFT analysis and objective analysis, to determine what system components should be selected for use in the theatre system design.

SO, WHERE'S STILLER AND MEARA

No trip to Casino Land would be complete without a jitney ride to the "Other Atlantic City," Harrah's Marina Hotel and Casino. After a five minute bone-shaking ride over to the marina section, we arrive at Harrah's. Our friend and compatriot Joe Marchione, the head sound technician, takes us on a grand tour of the Broadway by the Bay Theatre.

This room is without a doubt the nicest in town, with over a half a dozen seating levels. Every seat has a good view of the stage. The maple stage floor is 70 ft × 47 ft. not including the wings and storage space. For anyone who has ever unloaded more than one truckload of equipment, the truck-sized loading dock at the rear of the stage is heaven sent. Harrah's has the only theatre with complete carpentry, scenery, and electrical shops adjacent to the stage. It is also the only complex with in-house ice equipment for preparing the stage for ice shows.

Right in the center of the house, 25 feet from the stage apron, is the spacious and neat sound booth. It houses the customary PM-2000 32×8, with a PM-1000-16 submixer, and five racks of signal processing, containing various UREI EQs and crossovers, Lexicon 200, AKG BX5, and a Yamaha E1010.

The house system is a four-way central cluster comprised of four JBL 15s in Stanley Screamer subwoofer cabinets, four Altec 817 mid-bass bins, two CL&S ABH-90 radial horns with JBL 2441s, two Altec MR-64s with 291s, and two CL&S SQ90 high frequency horns with 808s. Joe presently has plans to install left/right stereo effects loudspeaker systems.

Monitor systems are always interesting, and this one is no exception. Starting with the console, the system consists of a Soundcraft 400B, EAW MH115 three-way side fills tri-amped with Crown PS-400s, all UREI EQs, Yamaha 2115s and 4115s, and PAS 12-inch and 15-inch coaxials. Joe says, "The performers and their soundmen love the PAS's, so we're in the process of switching over to them."

Harrah's has staged a full range of shows from Peter, Paul & Mary to Vegas-type stage/revues to the Beach Boys. Several rock acts, before they came to Harrah's, said that they would have to use their own sound systems; once they saw the system and heard its capabilities, they were happy using the house equipment.

With Harrah's elaborate stage-rigging and flying systems, a GE 5050B light valve projector (projects any video signal onto a movie size screen), theatrical lighting, and TV lighting systems, they have had the opportunity to work a full spectrum of special events, including video-tele-conferences, horse auctions, car shows (with cars on stage), and televised sporting events.

Joe comments on one of the more interesting events, The March of Dimes Telethon. "We had a full orchestra, announcers, lots of different acts, full communications, heavy individual mic'ing, and a full split to the outside video truck. The challenge was that we had the Tony Orlando Show that evening, and had to change the stage. That took a lot of planning and teamwork."

Joe explains how Harrah's policies reflect the way the audio department operates, "A lot of soundmen have a preconceived notion, based on bad experiences, that when they walk in the door they're going to run into an uncooperative house. They have so often been told, 'This is the way we do it,' or, '60 Hz hum? Problem? What problem? That's the way it always is....' We try to turn that attitude around and be as helpful as we can. Bill Harrah always did everything with style and treated everybody first class. We're happy to be carrying on those traditions."

After cashing in our chips and using up our last comp (complimentary tickets for food, drink, entertainment, etc.), we hit the road for the real world and left this Land of Oz behind us. We've seen that the role of the audio engineer in Atlantic City has three key aspects: art, science, and business. To quote The Great Yellow Book[3], "Having worked out the distribution, and smoothing off the house curve, you are then left with the actual problems of the microphone response.... There is also the irrational to consider—when the performer prefers a certain shape, size, color, trade name, or personal microphone." The Davises are pointing out that no matter how technically competent you may be, you have to be prepared to diplomatically deal with performers' "irrationalities."

Just as hearing is one sense that contributes to the overall perception of an event, seeing that "picture" is equally important, and there must be coordinated efforts by the stage, light, and sound crews to create the desired effects. When it all comes together, the audience is truly captivated by their Atlantic City experience. ■

3. Davis, Carolyn and Don. *Sound System Engineering*. Indianapolis: Howard W. Sams, 1975.

FULL RANGE SPEAKER SYSTEM

• Cerwin Vega's newest full range speaker system, the V-40, consists of Cerwin Vega's B-215 (direct radiating, twin 15-inch bass) combined with an RMH-1000 (fully encased JMH-1 high compression driver with aluminum radial horn). The system does not require bi-amplification to achieve maximum performance. Two V-40 systems can be driven by a single, dual-channel amplifier such as the Cerwin-Vega A-600 (600 watts-per-channel, 4 ohms). The system's direct-radiating design assures alignment of acoustic sources so that sound energy is propagated as a single wavefront, without smearing over a period of time. As a result, reproduction of percussive and other similar transient sounds are well-defined while music and vocals take on a greater realism and more natural sound. The V-40 crossover is a high-quality, passive 1 kHz, 12 dB-per-octave high pass filter that is optimally matched to the system components. The network was designed for low loss, high power handling, low distortion at high input levels, and to preserve dynamic range. The enclosures are fitted with recessed handles, metal corners, and rubber feet for ease of handling and setup. The computer-designed B-215 enclosure provides the internal volume and porting necessary to provide an optimally flat (fourth-order Butterworth) bass response characteristic with a 3 dB down point at 35 Hz. The drivers operate in individually tuned enclosures with closely coupled vents for maximum efficiency. In the unlikely event of driver failure, the remaining driver will operate with performance characteristics essentially unchanged with only a 3 dB reduction in SPL. The performance characteristics and cost effectiveness of the V-40 make the system eminently suited for use in portable and fixed installation sound reinforcement, high-quality music playback, wide-band cinema sound reproduction, and side-fill stage monitoring.

Mfr: Cerwin-Vega
Price: $940.00 (Suggested Retail)

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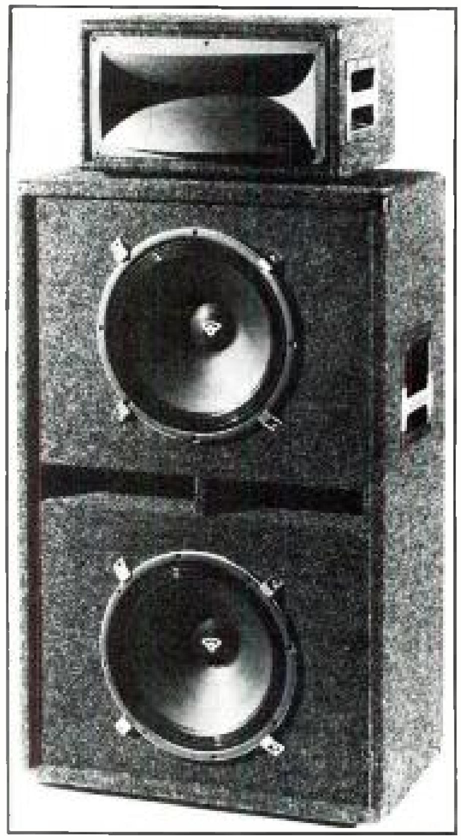
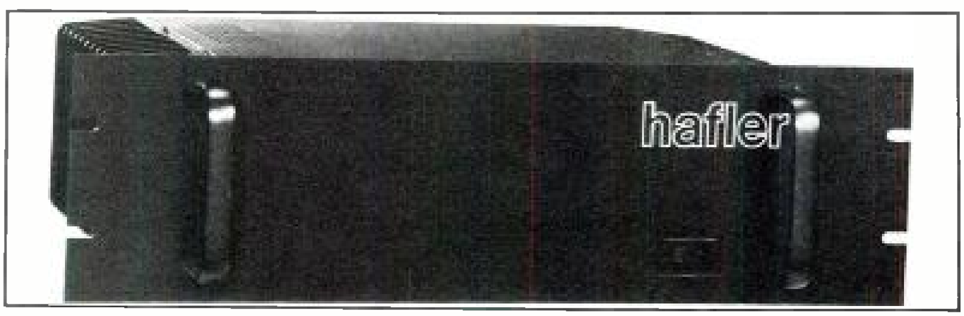


NEW POWER AMPLIFIER

• The David Hafler Company's P225 professional power amplifier is conservatively rated at 175 watts-per-channel into 4 ohms with less than 0.03 percent THD over the frequency range of 20 Hz to 20 kHz. It uses a push-pull complementary symmetry circuit which employs MOSFET output devices and quality components throughout, eliminating the need for complex and sonically degrading current-limiting circuitry. Input connections are via ¼-inch phone jacks, while output connections are heavy-duty five-way binding posts, spaced on standard ¾-inch centers to accommodate dual or single banana plugs. The P225 employs rear panel output fuses for load protection as well as thermal circuit breakers mounted to heat sinks. The thermal breakers will shut the amplifier down in the unlikely event of overheating. The P225 can easily be converted to a monophonic amplifier by its internal mono/stereo switch. In the mono mode, it delivers 350 watts into an 8-ohm load. Input gain controls on the amplifier's rear panel make level matching possible with a simple screw driver adjustment. Optional accessories include a 70-volt line transformer, a differential balanced/unbalanced input control board, and a multi-voltage transformer. The amplifier in its standard configuration is capable of operating from a 120-volt, 60 Hz AC line, and is provided with a three-wire grounded AC power cord. Its dimensions are 19-in. wide, 5¼-in. high, and 10½-in. deep. It is finished in black and is available either fully or partially assembled.



Mfr: The David Hafler Co.

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DIRECTIVITY CONTROL HORNS

• Community Light & Sound's new directivity control horns provide uniform horizontal and vertical coverage throughout their operating range. The PC294 (90 by 40), PC264 (60 by 40), and the PC242 (40 by 20) horns take a two-inch throat, four-bolt driver such as the JBL 2441 or TAD 4001. On- and off-axis frequency response of these horns is consistent from 500 Hz to 16 kHz. These black fiberglass horns are smaller than most available constant directivity horns, making them extremely useful in installations of limited space.

Mfr: Community Light & Sound

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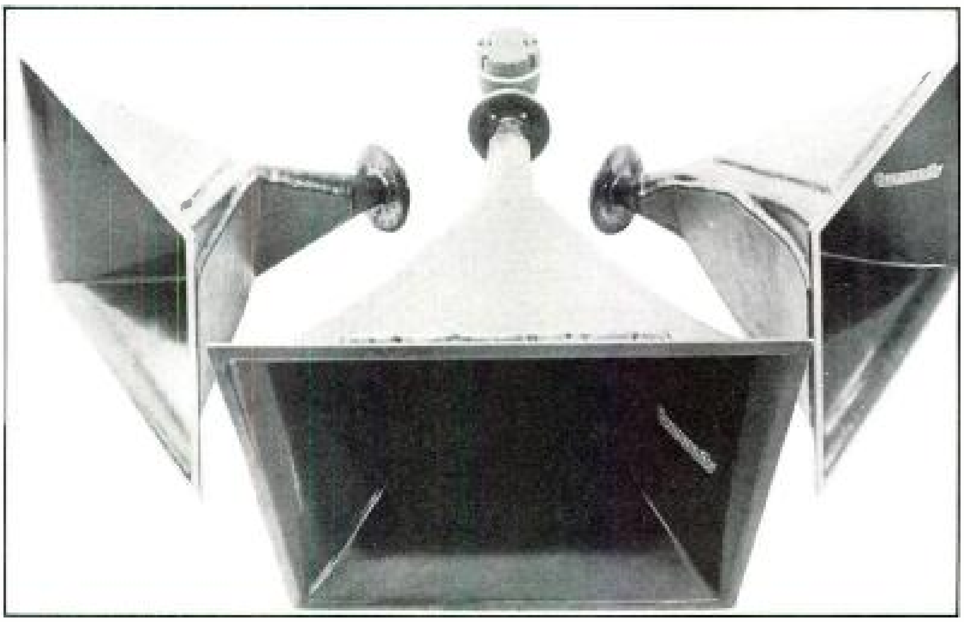
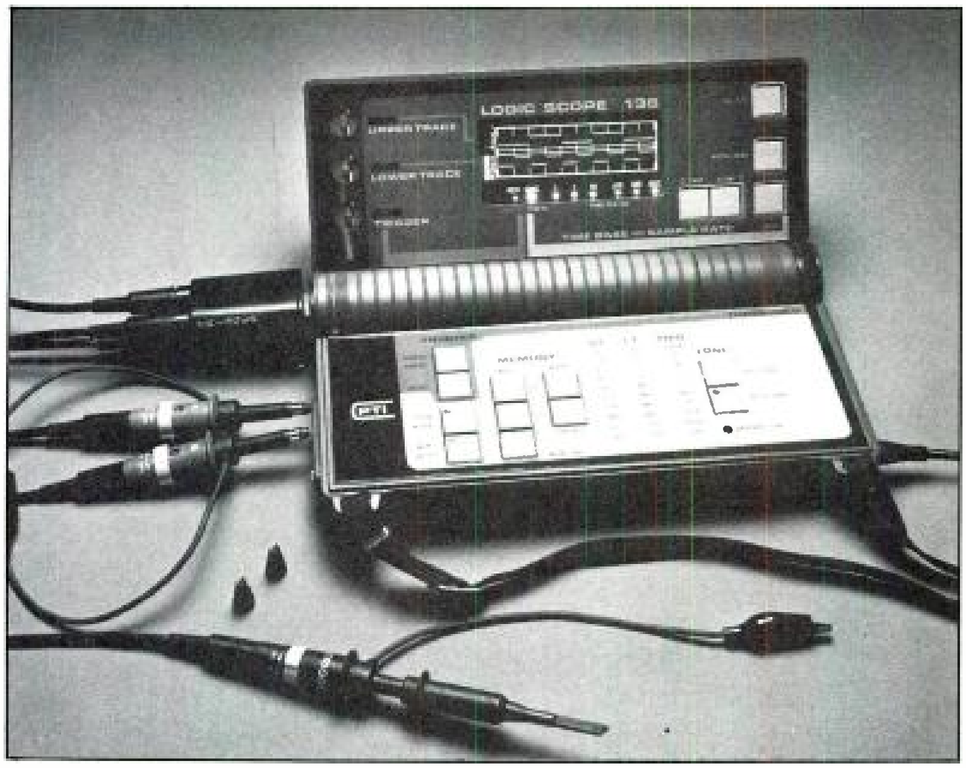


THE POCKET LOGICSCOPE



• The Pocket LogicScope 136, new from Pocket Technologies, Inc., is a test instrument that weighs less than one pound, operates on a 9-volt battery, and fits into a pocket. It has a true dual trace and conventional oscilloscope features including trigger, memory, single sweep, free run, reset, recall, and write. It also has conventional logic analyzer features including logic compare, "and," "or," and "exclusive or." The Pocket LogicScope 136 measures pulse widths down to 10 ns, detects logic glitches between 50 and 100 ns, and has a 10 MHz visual bandwidth and 20 MHz logical bandwidth. 16k of CMOS memory provides storage for up to 16 waveforms as well as memory link mode; an Input/Output connector allows waveforms to be sent to external systems for analysis/display. Other functions include "Auto Seek," which automatically adjusts the timing to fill the display with one complete cycle, and "Audo Trak," which allows the user to "listen to logic" with dual tone audio capabilities to measure logic states. Accessories included are a neck strap, carrying case, triple probe/cable assembly, and BNC adapter.

Mfr: Pocket Technologies, Inc.

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LOW FREQUENCY DRIVERS

• Electro-Voice's DL series of low frequency drivers, designed for high level, high quality sound reinforcement, playback, and studio monitoring, now includes two new models—the DL12X and DL18X. Like their predecessor the DL15X, these models perform at the high efficiency end of the direct radiator spectrum (on the order of five percent). They've been designed with different bass responses, enclosure size requirements, and target applications. According to EV, the woofers' carefully engineered drive system assures high efficiency, linear, low distortion output, and high power capacity. The low mass voice coils are made of rectangular aluminum wire, edge-wound on a rugged laminated polymide form. A break-up-resistant diaphragm and suspension ensure a smooth, musical upper-bass sound and plenty of low frequency shock capability or "punch." The DL12X driver is suited to midbass applications in three- or four-way systems or as a woofer in two-way systems where response to 70 Hz is appropriate, and a small enclosure, such as the 1.2-cubic-foot Electro-Voice TL806, is required. Its specifications include a frequency response (in a 2.6-cubic-foot vented enclosure tuned to 52 Hz) of 58 to 5200 Hz ±3 dB, a 350-watt long term average power capacity per AES recommended practice (100 to 1000 Hz), and a sensitivity of 100 dB (average from 200 to 4000 Hz) at 1 meter, 1 watt. The DL12X weighs 18 lbs. Designed for subwoofer use in three- or four-way systems, the DL18X offers the highest output in the lowest octaves because of its large cone area. Specifications include a frequency response of 36 to 3000 Hz ±4 dB, a 500-watt long term average power capacity per AES recommended practice (100 to 1000 Hz), and a sensitivity of 99 dB (average from 200 to 4000 Hz) at 1 meter, 1 watt. The DL18X weighs 21 lbs. Electro-Voice also offers plans for computerized vented enclosures and horn enclosures.

Mfr: Electro-Voice

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BATTERY-POWERED FREQUENCY COUNTER

• Leader Instruments' new LDC-831 is a 150 MHz, 4½-digit battery-powered frequency counter. The small size and portability of the LDC-831 make it well-suited for field service applications; an optional AC adapter converts it for bench use. Simple operating controls and a large, bright display of green LEDs make for easy operation. For high frequency applications, the LDC-831 can provide up to 6½ digits of resolution. The five most significant digits are displayed using the .01-second gate time. By then switching to the 1-second gate, all four digits to the right of the decimal point are displayed. The combination of the two readings yield 6½ digits of resolution. The basic accuracy of the LDC-831 is ±10 ppm. The unit carries a standard two-year warranty.

Mfr: Leader Instruments Corporation
Price: $210.00 list

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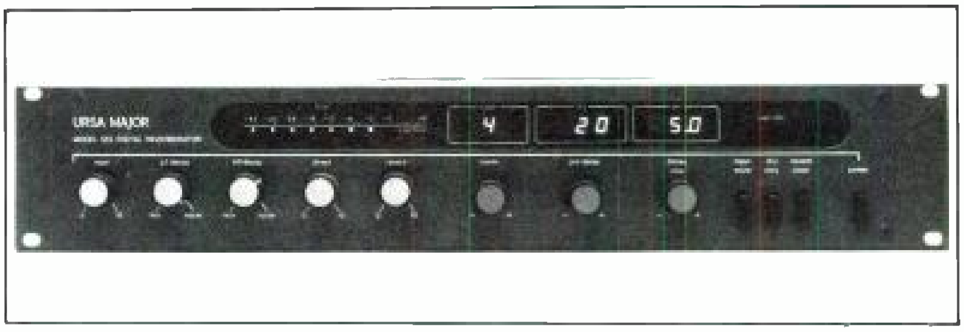

STEREO DIGITAL REVERB

• Ursa Major, makers of the 8×32 Digital Reverberator and the Space Station SST-282 digital effects unit have announced the third product in its line—the StarGate 323. The StarGate 323 is a stereo digital reverb for professional applications. The StarGate 323 is an extremely flexible device with eight room-simulations including tiny chambers, fast-diffusing plates, concert halls, and huge echoing spaces. The rooms can be modified by front-panel controls that adjust decay time, pre-delay, and separate high and low frequency decay curves. The unit is user-friendly, so the operator whose experience extends only to conventional plate and spring reverbs will understand its operation immediately. Digital readouts on the front panel show decay time, pre-delay, and room, while eight discrete LEDs monitor signal level. There is also an input level control, input-mute, reverb-clear, and dry-only buttons, each of which can be operated by foot pedals. Engineers familiar with first-generation digital reverbs will be able to use the 323's advanced features to their full advantage. The unit features a 15-kHz bandwidth and a dynamic range of 80 dB.

Mfr: Ursa Major

Circle 52 on Reader Service Card

HIGH SPEED CASSETTE COPIER

• Telex Communications' new ¼-track, 4-channel stereo model has been added to its popular Copyette™ line of high-speed audio cassette copiers. The new Stereo Copyette has a tape speed of 21 ips (11:1 speed ratio) and will automatically copy all tracks and both sides of a cassette in a single pass, thus achieving a copying ratio of 22:1. For example, a one hour (C-60) cassette could be copied in less than three minutes. The unit is simple to operate and requires only one lever action to copy tapes. When the copy is finished, it rewinds the tapes back to the beginning and shuts off automatically. The copier weighs 12 lbs. and is housed in a portable case. The new Stereo Copyette is ideally suited for church, school, industrial, and personal applications where good quality stereo sound duplication is required at a low cost. It utilizes DC servo drive motors that maintain a constant running speed and remain cool and stable throughout the copy cycle.

Mfr: Telex Communicators, Inc.
Price: $799.00 (suggested retail)

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4. Noise Control. *M. Rettinger*. Revised and enlarged into a separate volume. Covers noise and noise reduction. measurement and control. Several graphs and charts. 1977. App. 400 pages. **$28.50**

• Beginning Friday, March 23, public radio station **WGBH** Boston will be transmitting, via digital audio, live broadcasts of the Boston **Symphony Orchestra** (BSO). According to **Anita McFadden**, operations manager of WGBH Radio, the station will use two **dbx** Model 700 Digital Audio Processors to send weekly Boston Symphony concerts direct from Symphony Hall to their transmitter site in nearby Milton, Mass. The station conducted unannounced tests of the system in December and January. The obvious improvement in audio resulted in immediate acclaim by WGBH 89.7 FM listeners and engineering colleagues from other Boston stations.

A team headed by **David St. Onge**, chief engineer of WGBH TV/FM, which combined the talents of engineers from both WGBH 89.7 FM and its sister television station, WGBH Channel 2, was organized to test the dbx system. The team set up microwave links between Symphony Hall and Boston's Prudential Center, where Channel 2 has a video repeater to the WGBH studios, since there is no line-of-sight path for a single "hop," according to McFadden. Another microwave path was established between the WGBH studios and their Milton-based FM transmitter site.

• **Eddy Offord Studios** has recently purchased a new 24-track Studer A800, making the facility the first in the metro Atlanta area to own Studer's top-of-the-line multi-track unit. Other new equipment acquisitions include a **Synclavier**, **Simmons** drums, and a **Lexicon** 224X digital reverb. Recent projects produced and engineered by Eddy Offord at the facility include LP sessions for **Blackfoot**, on **Atco** Records, and **Pallas**, a popular English band recording for **EMI**. **The Police** were also in the studio to do overdubs and mixing for their upcoming Showtime video special. Other artists produced and engineered by Offord in the past include the **Dixie Dregs**, **John McLaughlin**, **Paul Butterfield**, **Yes**, and **Emerson, Lake and Palmer**. Offord's unique studio is located in the former East Point Theater, originally built in 1940. The recording console is located in the orchestra pit area, so there is no physical separation between the console and the on-stage recording area.

• **Broadcast Technology's** president, **Louis F. Lindauer**, has announced the appointment of John J. Bubbers as director of Marketing. John Bubbers has been associated with the audio industry for many years. He has been a broadcaster and a manufacturer of phonograph records and electronic equipment and has been associated with standards groups, including the 1964 NAB Disc Playback Standard, the IEEE Standard on Measurement of FM Tuners, and the IHF Standard for the Measurement of Amplifiers. He is a Fellow of the **Audio Engineering Society**, a past President and its current Secretary, and a Bronze Medal recipient. He is also a member of the **IEEE**.

• **Cetec Gauss**, manufacturer of tape duplicating systems, has incorporated and introduced the **Dolby** HX Professional biasing device in its Series 2400 high speed cassette duplicating systems. The company also announced that Dolby HX Professional biasing technique, which can significantly improve the audio quality of cassette recordings, is being offered as a factory option conversion kit for existing Series 1200 high speed duplicators. For the professional duplicator of prerecorded cassettes, HX Professional biasing technique improves the high frequency performance of cassettes with prerecorded programs which have high frequency, high energy content.

The HX Professional system is supplied with two preset positions for ferric oxide and chromium dioxide tapes. A third position is provided for discriminating users adjustable for the utilization on more exotic tapes, such as metal particle tape. The speed control switch will automatically switch between 64:1 and 32:1 duplication ratio. The HX Pro can be used on more than one duplicating speed. There is a switch on each "slave" of the duplicating system to turn "on" and "off" the HX Pro biasing technique. To assist in production operation, an operator has the capability of switching the HX Pro "on" or "off" without any system down time or set-up changes.

• **Harro Heinz**, president of **Renkus-Heinz, Inc.**, has announced the appointment of **Gregg Wilson** to the position of director of Sales and Marketing. Renkus-Heinz, of Irvine, California, manufactures a full line of professional audio components including drivers, horns, crossovers, and speaker systems. As director of sales and marketing, Wilson's duties will include coordination of Renkus-Heinz' internal and field sales activities, continuing market development, and promotion and advertising of the company's products. Wilson comes to Renkus-Heinz from **Fender Musical Instruments**, where he held the position of district sales manager in the Pacific Northwest.

• **The Peabody Conservatory of Music**, America's first established music school, has recently entered into a unique commercial venture with a dynamic young recording company, **Sine Qua Non Productions**. Over the next five years they will issue a series of thirty digital recordings featuring the stellar talent of Peabody faculty, alumni and students. The recordings, packaged with all the artistic flair that has become a trademark of Sine Qua Non, will be marketed nationally through record stores and through such non-traditional outlets as book and audio stores. Peabody's insistence on the highest quality of music and audio performance, coupled with Sine Qua Non's reputation for artistic selectivity, means that the recordings are planned to have a long life in the marketplace. The suggested retail price of about $8 per cassette or record will make them extremely attractive to the consumer.

Among the many unique features of this undertaking is the fact that the releases will be engineered and produced in Peabody's new state-of-the-art recording studios under the direction of faculty member and recording engineer **Alan Kefauver** and involving students in the school's Bachelor of Music in Recording Arts and Sciences degree program. The degree program, inaugurated in the Fall of '83, is the most rigorous in the country and aims to train a new breed of highly sophisticated audio engineers/producers. Analogous to the "Tonmeister" program run by a consortium of European broadcasting companies, the five-year degree program is made possible by Peabody's affiliation with **The Johns Hopkins University** and is based on the joint resources of the conservatory and the university's G.W.C. Whiting School of Engineering.

Among the first faculty artists to record in the series will be pianist **Ann Schein**, cellist **Yehuda Hanani**, and the **American String Quartet**, Peabody's Quartet-in-Residence. It is anticipated that these initial recordings will be available by late Spring. **Leon Fleisher** is among the Peabody artists who have agreed to record at a later date.

Sine Qua Non Productions, founded in 1971, has made its mark in the industry and the marketplace. Headquartered in Rhode Island and operating out of a renovated brass foundry off Providence's historic Moshassuck Square, the firm's proximity to the **Rhode Island School of Design** has given it a rich creative source for imaginatively designed covers. Among the artists and ensembles that have appeared on its labels are: **Laurindo Almeida, Enrique Batiz, Emanuel Borok, Philippe Entremont, Morton Gould, The London Symphony Orchestra, Jean-Pierre Rampal**, and **Peter Serkin.**

• **Fender** has announced the appointment of **Drew Daniels** to the newly created position of Sales Training manager. Daniels' chief responsibilities will be coordinating a new program of sales training seminars for Fender dealers and producing materials necessary for the program. The seminars will be conducted in the new multi-media sales training facilities at the Fullerton plant. Daniels is currently serving a term as chairman of the Los Angeles section of the **Audio Engineering Society** following a term as vice-chairman and three terms as treasurer. Prior to his association with Fender, Daniels served as a technical specialist at **Teac/Tascam** and **JBL Sound, Inc.** where his responsibilities included production of technical literature, the design of electronic interface circuitry, TDS testing and the training of sales and laboratory personnel. In addition, he has held the position of systems engineer on a variety of projects including the design of recording studios, installation of a Busch Gardens special attraction sound effects system, and the installation of the sound system at Los Angeles' Dodger Stadium.

• **Unique Recording** has recently updated all 14 of their digital and analog polyphonic synthesizers and sequencers to MIDI (Musical Instruments Digital Interface) spec 1.0. MIDI allows a musician to control any number of MIDI-equipped synthesizers simultaneously from any MIDI-equipped keyboard, guitar or synthesizer. An example would be: 10 eight-voice synthesizers (16 oscillators each) allowing a musician to have control of 160 oscillators from one keyboard. This wall of sound up to now has only been available to synthesists with large elaborate modular systems such as artists like **Tomita, Keith Emerson** and **Wendy Carlos.**

Unique has also added an **AMS** DMX 15/80S stereo harmonizer/delay for rental, and a second **Sony** DRE 2000 digital reverb for Studio B, a second **Yamaha** DX-7, a **Commodore** SX-64/SCI MIDI sequencer, and a **Roland** TR909 drum machine.

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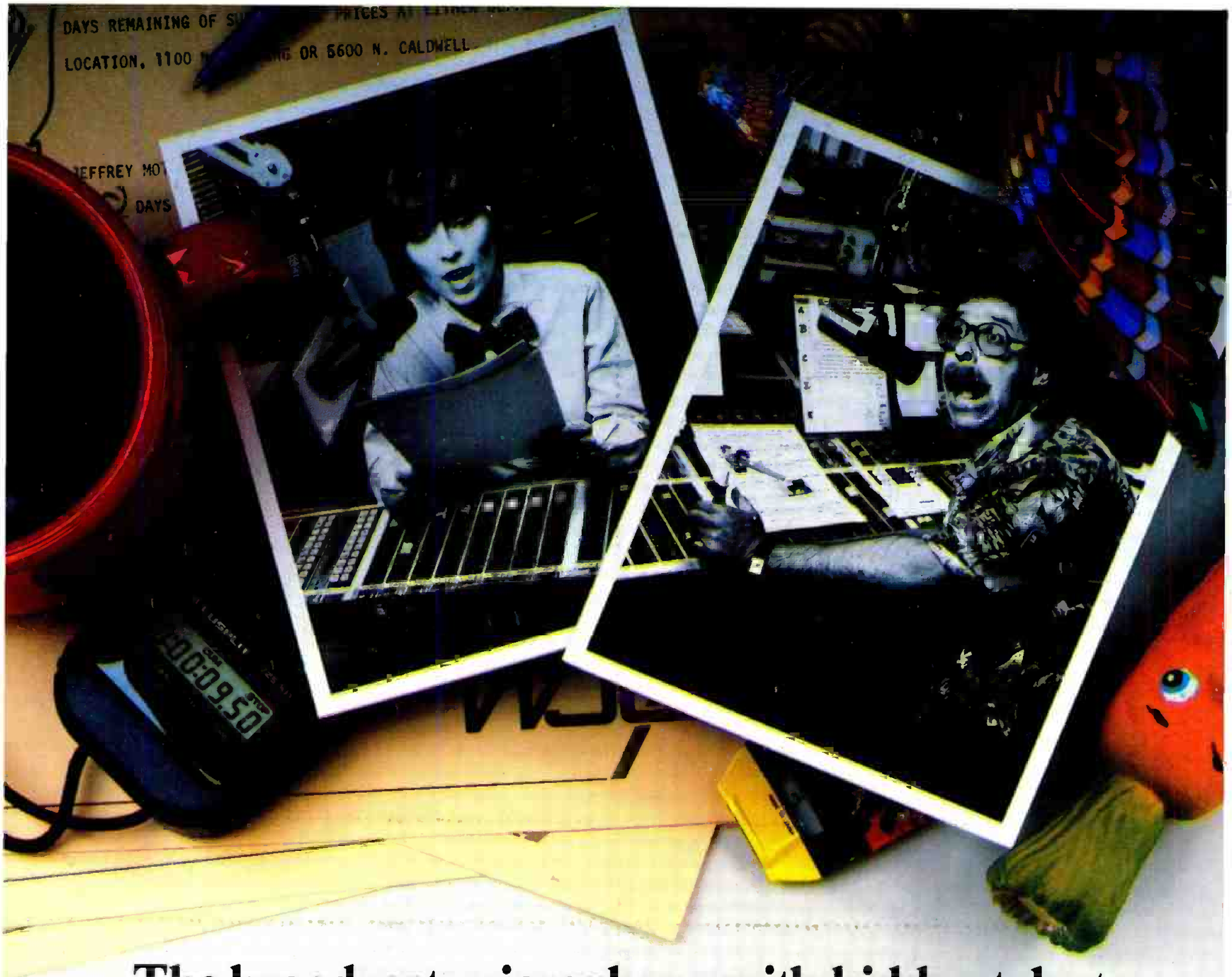

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To help broadcast engineers contend with a wide variety of voices, the Shure SM7 is really four microphones rolled into one. That's because the SM7 Unidirectional Dynamic Microphone features two frequency-tailoring switches that provide a choice of four different response curves—to best suit each individual voice and situation.

Depending on the switch settings, the SM7 can provide an extremely wide-range flat frequency response, add presence and crispness to speech, boost vocal clarity, roll off low frequencies to provide natural closeup miking, or help reduce sibilance.

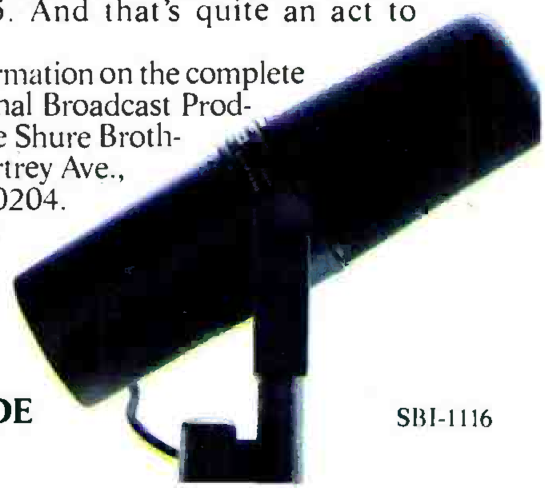
At the same time, a tight cardioid pattern effectively rejects unwanted background noise and minimizes off-axis coloration.

Beneath the SM7's integral foam windscreen is a rugged steel cage that surrounds and protects the cartridge from damage. And Shure's patented air suspension shock mount offers uncompromising isolation, cutting down on the effects of mechanical vibration in the studio.

Engineers will also appreciate the built-in humbucking coil that guards the SM7 against electro-magnetic interference.

With so many talents, it's no wonder the SM7 is following in the successful footsteps of Shure's legendary SM5. And that's quite an act to follow.

For more information on the complete line of Professional Broadcast Products, call or write Shure Brothers Inc., 222 Hartrey Ave., Evanston, IL 60204. (312) 866-2553.



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