THE SOUND ENGINEERING MAGAZINE

FEBRUARY 1982
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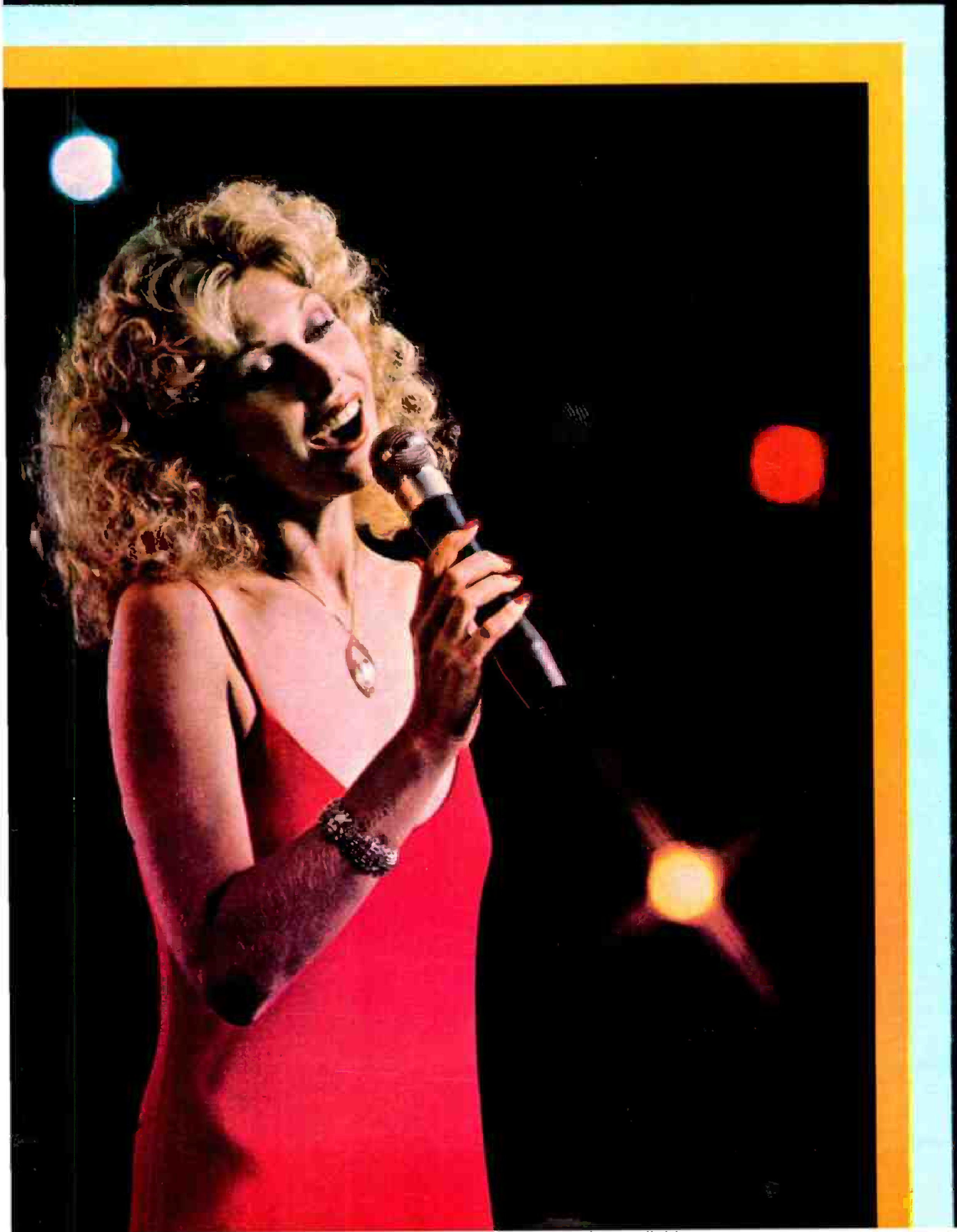


Beginning as a Nashville session musician with a burning desire to be a producer, Larry Butler watched and listened. His first break came when he got a producer job with Capital Records in Nashville. The first record he ever cut, with Jean Shepard, was a hit. Since then he has cut over 50 gold and platinum records as producer for CBS, Johnny Cash Productions, Tree International, United Artists and now as an independent. His recent relationship with a man named Kenny Rogers, has produced hits like Lucille, She Believes In Me and The Gambler. Larry won the Grammy Award as producer of the year in 1980.

ON DEVELOPING A STYLE

"When I started producing, I was producing like everybody in town. I started to produce a record like Billy Sherrill would do it or like Owen Bradley would do it or whatever. And then one day I listened to a lot of records I had done and I thought now wait a minute. If somebody wants a record that sounds like a Billy Sherrill record they can go get the real thing. So I started producing the way I wanted to produce. It was a great lesson for me. It was a big turning point in my career. I think that nobody is really going to sell or really succeed until they reach that point where they're putting themselves into it, instead of making a copy of someone else's work."

ON REACHING THE LISTENER

"I'm a believer in the simplicity of a song. I believe in laying something in somebody's lap they don't have to search for mentally. I've said this before, if a guy's driving home from work he's got a million things on his mind. He's got to spank the kids when he gets there. He's got a flat tire on the way home. And through all of this there's a song. He's got his radio turned down kind of low and a song cuts through all of that and he finds himself humming along with it. When that happens you've hit one in the upper decks."

ON KENNY ROGERS

"Kenny is such a universal name, such a big name. I try not to let any prejudice enter into comments about Kenny because we've been so close, but I guess he has to be the strongest single male artist in the United States. I can't think of anybody that's reaching the mass of people that he's reaching and I think it's unfair that people say he's the new Elvis. Well, there's never going to be another Elvis. There's Elvis Presley. That's it. Forever. But as far as sales, you might compare them."

ON KNOWING WHEN TO STOP

"I think the most common mistake for an engineer and producer to make is maybe not really realizing the take when they've gotten it. Sometimes going too far because they're looking for that emotion or magic. Sometimes you can have it and not realize it. Sometimes you can have maybe one guitar part that bothers you, so you go ahead and do another take. Well, you have gone by the one that had the feeling, the one that had the emotion."

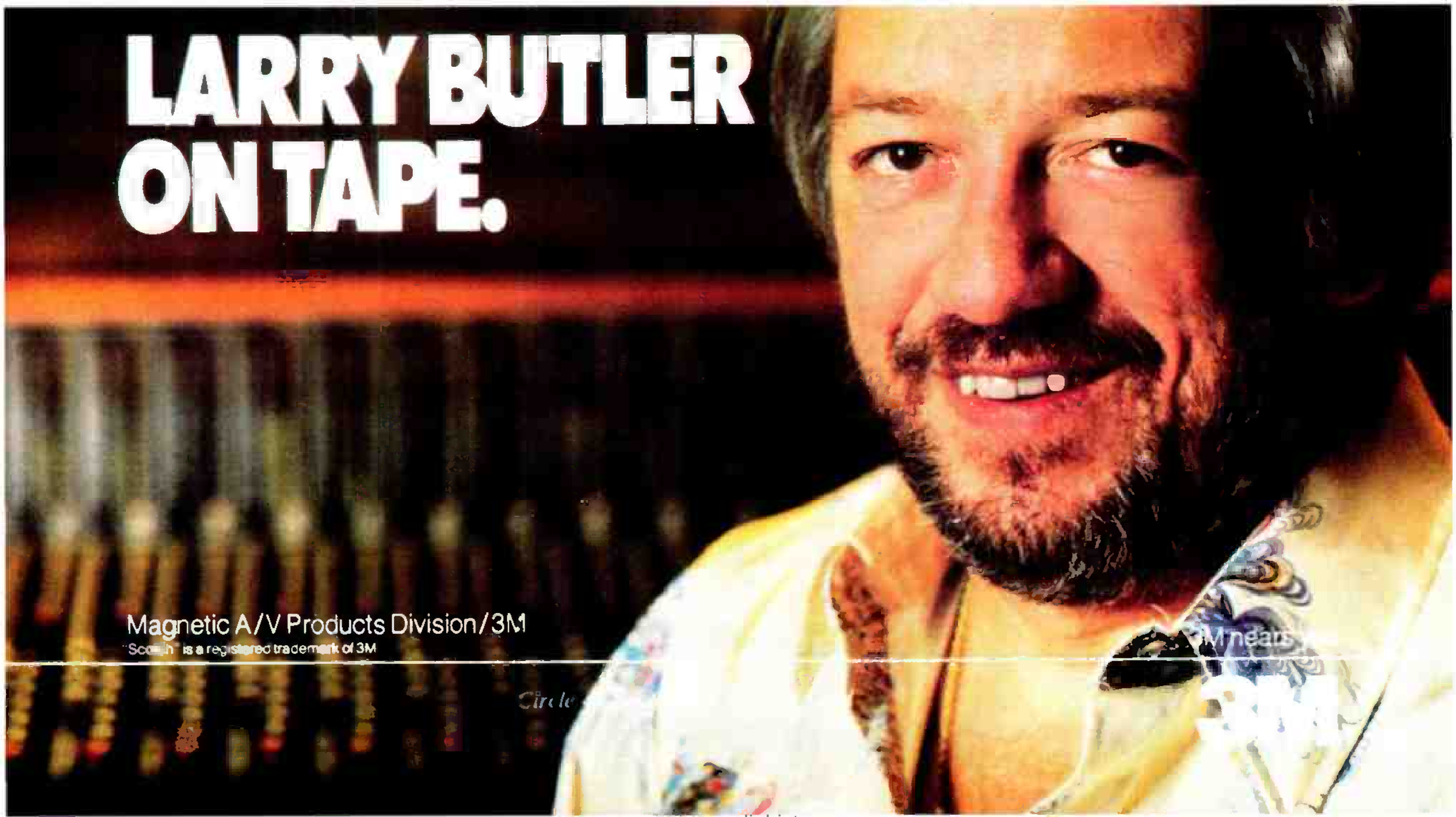
ON TAPE

"I use the philosophy and theory of surrounding myself with people who know what the hell they're doing and letting them do it. I let the engineer do his job.

The only things I've heard them say about 3M is it's dependable, you can trust it, you don't have to worry about it. When you're spending money and you get good service you're not going anywhere else. You're going to stay there with whoever it is.

I just know 3M has always been very, very open for ideas and suggestions. It's just like "money making music." Three M's. That's the way I think of the tape, because it works and it sounds great."

SCOTCH 250
WHEN YOU LISTEN FOR A LIVING.



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FEBRUARY 1982 VOLUME 16,NO. 2

DEPARTMENTS

FEATURES

About The Cover

• This month's cover features the Cetec Vega Model 80 hand-held wireless microphone highlighted in a concert situation.

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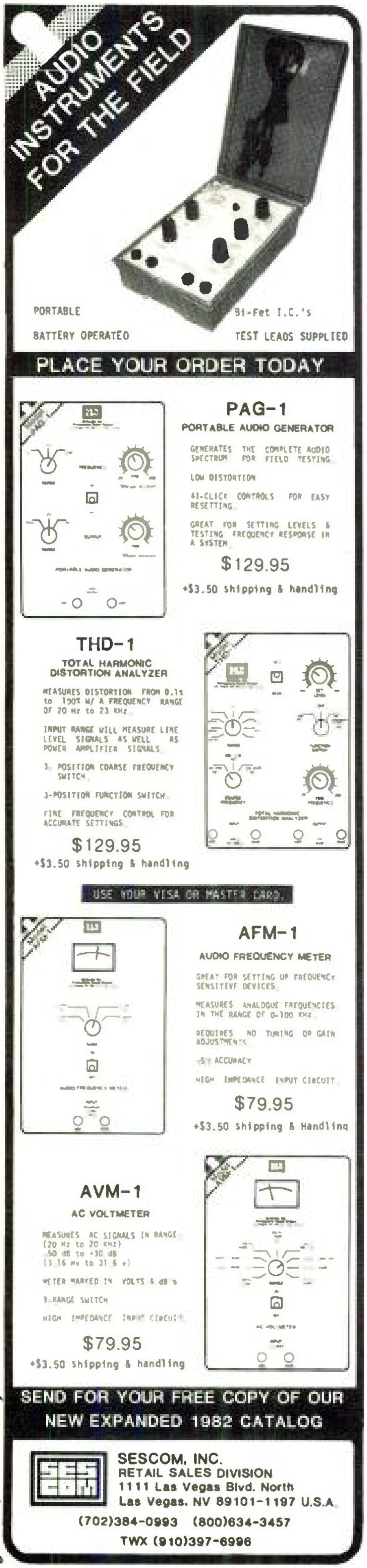

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db Letters

TO THE EDITOR:

Imagine my surprise to open the October issue of **db** and find a five-page advertisement for an acoustical product disguised as editorial material ("The Saga of Sonex" by C. Nicholas Colleran). Even your acknowledgement of the author admits that he is a distributor of the product.

Please don't get me wrong. I think sonex is a good product, and certainly one which people in the recording industry should be familiar with. But it is *not* the only product which can supply absorption in a studio. Nor is it fair to say that it "is the most cost-effective material for the uniform absorption and diffusion of sound."

In fact, the configurations of the Sonex foam are effective at increasing absorption and/or providing diffusion only at high frequencies, where the dimension of the wedges is equivalent to a quarter wavelength (or more) of the sound. Three-inch wedges, for example, would be ineffective below 1125 Hz. For this reason, the configurated foam is actually less efficient than solid, flat foam of equal thickness at low frequencies.

I am also at a loss to explain why the tabulated sound absorption test data does not correspond to the graphically presented data. At low frequencies, the tabulated data has much poorer sound absorption coefficients than the graphical data. And is it not a bit unfair to compare this product to an acoustical ceiling tile having an NRC of less than 0.45? Many acoustical ceiling panels today have sound absorption coefficients which are twice those shown.

I want to repeat my earlier statement. Sonex is a good product, with numerous applications, both in the recording industry and elsewhere. But it is not the far-and-away favorite, as implied in the article. And a magazine of the caliber of **db** should not be endorsing any product under the guise of editorial copy.

ERIC NEIL ANGEVINE, P.E.

TO THE EDITOR:

Thank you for giving me the opportunity to reply to Mr. Angevine's letter. I will ignore the comments referring to my article as a disguised advertisement. It should be obvious to anyone who reads **db** and other trade journals that the prime source of information on new technology comes from engineers or marketing people employed by manufacturers who have a direct interest in product sales.

My statement on the cost-effectiveness of Sonex was based on our having built several studios. I am a certified public accountant and that statement was a professional opinion and, as such.

Index of Advertisers

Coming Next Month

• In March, our topic is International Audio. Almon Clegg brings us up-to-date on the impact of Japan, Inc.. Richard Koziol checks in with a report on the Bach Madeira Festival and Curtis Chan gives us a look at Sony's 3324 Digital System.

In addition, noted author John Eargle joins our staff of columnists with a new column devoted to sound reinforcement. All this and more in **db—The Sound Engineering Magazine.**

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could be argued all day without resolution.

The sound absorption coefficients for Sonex were given to us by an independent laboratory, INTEST, in Minneapolis. The reports show both 3-in. and 4-in. Sonex to have an absorption coefficient of 1 at frequencies of 500 Hz and above. I would be interested in seeing any conflicting tests which would support Mr. Angevine's statement that the foam is ineffective below 1125 Hz.

It is quite correct that the tabulated data doesn't match the graph shown in the article. The graph was from test data done in accordance with ASTM 66. (The tabulated data is from ASTM C 423-77, as clearly indicated in the article.—Ed.)

With regards to the last paragraph, it may just appear to us that Sonex is the "far-and-away" favorite due to the difficulty we experienced last year in meeting demand.

C. Nicholas Colleran
President & Treasurer,
Alpha Audio

db Calendar

February

23-25 **Syn-Aud-Con Workshop:** Loudspeaker Array Design Workshop. San Juan Capistrano, CA. For more information contact: Syn-Aud-Con, P.O. Box 669, San Juan Capistrano, CA 92693. Tel: (714) 496-9599.

24-26 **Electro-optics, fiber optics and lasers for non-electrical engineers.** Given by George Washington University, Washington, D.C. For more information contact: Director, Continuing Engineering Education, George Washington University, Washington, D.C. 20052. Tel: (800) 424-9773.

26-27 **Country Radio Seminar.** Opryland Hotel, Nashville, TN. For more information contact: Dennis Buss, The Organization of Country Broadcasters, P.O. Box 120548, Nashville, TN 37212.

March

2-5 **71st AES Convention.** Maison des Congres, Montreux, Switzerland. For more information contact: Audio Engineering Society, 600 E. 42nd St., Rm. 2520, New York, NY 10165. Tel: (212) 661-8528.

6 **A Day With Syn-Aud-Con.** Montreux, Switzerland. For more information contact: Syn-Aud-Con, P.O. Box 669, San Juan Capistrano, CA 92693. Tel: (714) 496-9599.

The "Right To Tape" Movement

• Never has a single issue aroused the passions of members of our industry to the extent that the recent court decision concerning home video tape recording has done. For the few who have not already been caught up in this debate or who haven't bothered to read a newspaper or trade paper over the last several months, a quick review is in order.

Some five years ago, when home video tape recording was in its infancy, Walt Disney Studios, its distributors Universal Pictures, and MCA (the parent of Universal) elected to sue Sony Corporation. They felt that Sony's Betamax home video recorder threatened to deprive copyright holders (writers, actors, artists, producers, etc.) of income to which they were entitled. Remember, all of this started long before the current surge of pre-recorded video tape popularity occurred, a new type of home entertainment program which is producing profits for those very copyright holders (including Walt Disney, Universal Pictures, et al) who object to people owning and using video recorders. But I'm getting a bit ahead of the story.

Citing an analogous precedent in the field of audio, the VCR makers (and litigant Sony) felt sure that the courts would rule in their favor and they mounted an impressive case before the courts. The precedent, incidentally, had to do with audio recording. In 1971, when the same sorts of issues were raised concerning private off-the-air or even off-vinyl recordings for *non-commercial* purposes, the Congress of the United States specifically exempted such in-home or private recording from the then-applicable Copyright Act of the U.S.

Sure enough, Sony, as well as the other "defendants" in the case (a Sony distributor, their advertising agency and even a test-case consumer named in the suit), emerged victorious as the Federal Court ruled that such non-commercial video recording did not constitute unfair use according to the Copyright Act. At that point, even though most of the industry forgot about the case, the plaintiffs appealed the decision, and everyone assumed that the Appeals Court would simply rubber-stamp the lower court decision. Such, however, was not the case and on October 19th the Ninth Circuit Court of Appeals overturned the lower court decision and ruled it illegal for consumers to video tape copyrighted programs off the air, even for private, non-commercial use.

HOW ABOUT VIDEO?

That brings us to the question of audio recording from FM or stereo FM radio, or the copying of records onto home tape. Once this was as "hot" an issue as the video decision we have been talking about. How come we haven't heard much about it in recent years? The answer has to do with a bit of confusion that exists in the minds of many who are concerned with these matters. Many journalists, in discussing the taping of audio programs and the question of possible illegalities connected with such activity, have pointed to a congressional act which was passed way back in 1971. That bill specifically excluded home video recording from the then-existing Copyright Law so long as the audio recording was not used for profit or sold. In fact, it was the precedent of that Congressional action which has prompted the video consumer electronic industry to encourage similar legislation that would bypass the Court entirely in the present instance.

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that the Copyright Laws of the United States were completely re-written in 1976 and the *new* Copyright Act of 1976 does *not* include or carry over the specific 1971 Congressional act which excluded home recording from records or radio from the act, making it legal. In other words, the legality of private copying of broadcasts and records onto tape is again an open issue and we can expect that record companies, seeing the action taken by the Appeals Court in the case of VCRs, may well institute test cases to determine just what the status of home recording is.

It is not my intention to take one position or the other in this instance. I am sure that each side in the video recording argument feels that justice is on his side. Makers of VCRs (as represented by the position taken by the Electronic Industries Association and others) point out that copyright holders are not, in fact, damaged by video taping off the air. To support that argument, they offer statistical proof that between 70 and 90 percent of all off-air video taping is for the purpose of so-called "time shifting" (viewing a program that might otherwise be missed—just once—and then using the tape over again) rather than for the permanent storage and repeated viewing of copyrighted material over and over again). That being the case, it can even be argued that the copyright holders *benefit* by the video taping. Jack Wayman, Senior Vice President of the Consumer Electronics Group of EIA, in his testimony before a Senate Judiciary Committee on November 30, 1981 said, "Time shifting increases broadcast audience size by allowing those unable to view a program to view it later. A report commissioned by the FCC to assess the effects on broadcasters of home video recording recognizes the benefits copyright owners reap from time-shifting:

> "Because both major rating services—A. C. Nielsen and Arbitron—now include an indication of VCR use, this time-shift phenomenon should actually be an asset to the networks and broadcasters. Shows that would have been missed can now be recorded for later viewing.... An audience that was previously unavailable to them (the broadcasters) is now viewing and the viewing is properly attributed in audience reports.
>
> "Any fair analysis of VCR usage leads to an inevitable conclusion: time-shifting, the principal use of VCRs, economically benefits rather than injures the copyright holders."

In other words, since the viewing audience is increased, the sponsors, broadcasters and producers can be induced by the copyright holders of the program to offer them a better financial "deal."

This is a cogent argument in the case of video, and it is entirely possible that the recent decision of the courts will be reversed once again, either by a re-hearing, or by the Supreme Court or, if that fails, by Congressional action. But how would the same argument hold up in the case of audio recording—off the air or from records to tape? It wouldn't! The person who makes a cassette tape of a record or of an FM program does not make that recording for the purpose of "time shifting." Very few cassette decks we know of (and even fewer open-reel decks) are equipped with programmable timers such as those found on virtually all VCRs. And, while it is true that most of these decks could be equipped with external timers, very few of these add-on devices are actually used, if we compare their numbers to the millions of tape decks now in use throughout the U.S.

With that argument out of the way, what other arguments will makers of audio recorders and/or blank tape have when and if major test cases are initiated against them? The EIA, in its testimony before the Senate Committee, cited what it thought are two more arguments in favor of the "right to tape" at home. One of these, that "an injunction or other relief would seriously damage a young, fragile industry (the video industry)" hardly applies here, since the tape recording industry is no longer "young" nor is it particularly fragile—with home taping equipment selling more vigorously in an otherwise depressed consumer audio market than any other type of sound equipment.

That leaves only a third argument, which *does* have merit. That argument maintains that the most recent ruling is a serious infringement on the rights of private citizens. As Wayman put it, "...its (the court's) ruling has had the sudden effect of turning law abiding citizens in almost three million homes into law-breakers...." That same argument could certainly be applied to audio recording if a similar court decision were to be handed down, except that we would be talking about a great many more than three million law-breakers.

While much of the downturn in industry record sales can be attributed to the economic conditions of the country as a whole, there are those who argue that record sales slumps are also the result of extensive copying of records by people who buy or borrow a single disc and then allow other friends to copy that disc for their own recording libraries. Having no statistics available on the direct effect of taping of records on national record sales, I cannot take an authoritative position on this issue. What is clear, however, is that the entire copyright issue has not been able to successfully keep pace, at the legislative level at least, with advances in technology. Book publishers might well accuse photocopy machine makers of being responsible for a drop in sales. The reasoning extends to creators of computer software (which can also be copied by private, knowledgeable programmers) and so on, ad infinitum. Some serious rethinking of copyright laws is called for here—rethinking that is fair to *all* sides, if that's possible. ■

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NORMAN H. CROWHURST

db Theory & Practice

Parameters & Viewpoints

• A long while ago, tests showed that the smallest amount of equivalent harmonic distortion that human hearing could detect was about 5 percent. Two suggestions have been made to explain this, along with the fact that today we "know better." The first is that loudspeakers had more than 5 percent distortion—often a lot more—while the measurements were made on *amplifier* distortion, completely ignoring what the loudspeakers were doing.

What that meant was that the loudspeaker distortion, which was not being measured, masked the amplifier distortion, which was being measured. So when amplifier distortion was reduced below that 5 percent figure, you couldn't hear the difference because of the loudspeaker distortion. That sounds like a reasonable explanation.

The second suggestion was that the 5 percent figure was obtained by measurements made before the days of negative feedback. Some observers asked, what could the advent of feedback have to do with it? To which, like most things, you might get at least two replies. One would say, "Distortion is distortion," and feedback may be able to make it more or less, but what difference would that make? While others would proceed to find out exactly what feedback did, with regards to distortion.

Then, of course, some theorized and some measured, or at least went into their theory a little more thoroughly. Feedback theory often uses some deceptively simple algebra. If the gain is "A," and the feedback fraction is "B" (the engineering books use μ), completing the feedback loop reduces gain by the factor (1 + AB), called the "feedback factor," which can be converted to dB, and called "dB feedback."

That's the first step. Now, go through the algebra again, to see what effect feedback has on gain fluctuations, frequency response, distortion, input and output impedance and so forth. Each run-through of the theory seems to show that the quantity being investigated changes by that same figure: the feedback factor. To summarize:

Negative feedback reduces distortion by the feedback factor. Frequency response is a little more complicated, because you cannot analyze that without making reference to phase, but it is a function of (1 + AB) when you recognize that this quantity possesses phase as well as magnitude, and thus is a vector. If you have negative voltage feedback, you decrease output impedance by the feedback factor, while if it is negative current feedback, you increase output impedance by the feedback factor.

FEEDBACK FACTOR

If the feedback is shunt injected at the input end, negative feedback decreases input impedance, while if it is series injected, it increases it, each way by the feedback factor.

As we have shown before, the simple theory uses the same "magic number" (even if really it's more than a number, being in fact a vector) for all these purposes. But in real life, it isn't as simple as that. One way of looking at it says: if you "use up" your feedback to change impedance (for which purpose the cathode or emitter follower is a classic case), you don't still "have it" to reduce distortion by the same amount.

While that rather sloppy statement expresses the situation, if you analyze the situation (whatever it is) more rigorously, you must concede to the theorists that *at any given instant* there is only one feedback factor attached to one particular circuit, or piece of circuit, for the various applications of the theory. Those words in italics make all the difference. Theory treats feedback factor as a constant, which it is not; if it was, we would not need feedback for any of those purposes!

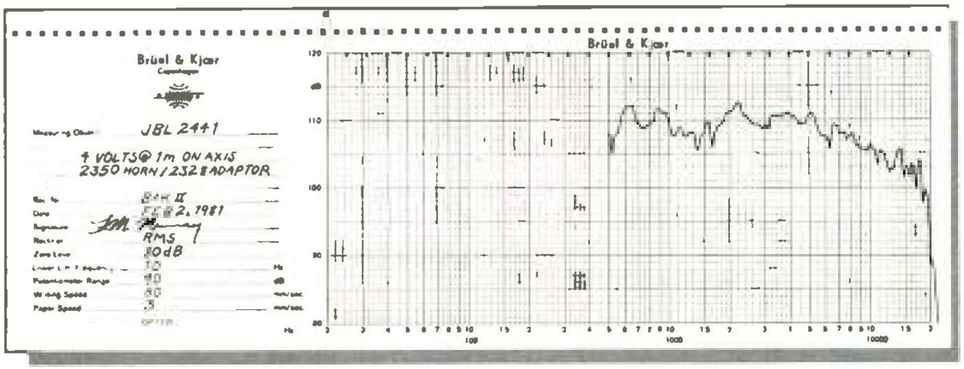
The feedback factor varies with frequency (including the fact that it must be considered as a vector for this purpose). It also varies with time—that is, during the progress of a waveform. So in the case of the cathode or emitter follower, if feedback serves to reduce gain by the gain of the stage, so that the resulting gain is unity (zero dB), then it reduces the output impedance, and increases the input impedance, by that factor.

But that factor is not constant. The figure we usually quote will be the average gain, over the signal used in that stage. It may be 100, or 1000 or some other figure. But it may vary at different points along the transfer characteristic of the device, say between 70 and 150. Feedback serves to "iron out" that variation, by reducing all of those values to unity.

Usually this means that impedances will be made so big, or so small, that fluctuations in their value will not matter, compared with other circuit values. For example, the input impedance will be more than 10 megohms, and the output impedance less than, say 10 ohms. If the source from which the 10 megohms draws its signal, or the one into which the 10 ohms feeds its signal, is 10,000 ohms, variations in those figures will not be significant. It won't matter if the input impedance runs from 10 to 20 megohms, or the output impedance from 5 to 10 ohms.

What feedback is doing, in this instance, is putting the effective values where fluctuation in value won't have any effect. But does this also happen to distortion? Let's take that example a little further. Every stage has at least one

This is not the only reason to buy JBL's new 2441 compression driver.



Unequalized frequency response of a typical 2441 on a JBL 2350 horn.

As you can see from this frequency response curve, the new JBL 2441 delivers impressive levels of performance. From extended bandwidth to high sensitivity and smooth, peak-free response.

But as important as these performance parameters are, they're only part of the story. Using the latest laser holography and computer analysis techniques, JBL engineers have developed a unique diaphragm design that allows the 2441 to match its outstanding response with unprecedented reliability and power capacity. That means you get exceptionally high performance without the trade-offs found in previous driver designs.

The secret behind this increased performance lies in the diaphragm's three-dimensional, diamond-pattern surround.[1] As outlined in a paper published in the Journal of the Audio Engineering Society,[2] this surround is both stronger and more flexible than conventional designs. This permits the diaphragm to combine all the traditional reliability and power capacity benefits of its aluminum construction with the extended frequency response of more exotic metals. It also maintains consistent diaphragm control throughout the driver's usable frequency range to eliminate uncontrolled response peaks.

Additionally, each 2441 is built to JBL's exacting standards. The magnetic assembly is machined from rugged cast iron and steel. Extremely tight machining tolerances and hand tolerance matching maintain unit to unit consistency. And finally, each 2441 is individually tested to ensure that it meets published specifications.

So before you buy any compression driver, ask your JBL professional products dealer about the 2441. It'll deliver a lot more than just an impressive frequency response.

1. Patent Applied For

2. Journal of the Audio Engineering Society, 1980 October, Volume 28 Number 10. Reprints available upon request.

James B. Lansing Sound, Inc.
8500 Balboa Boulevard,
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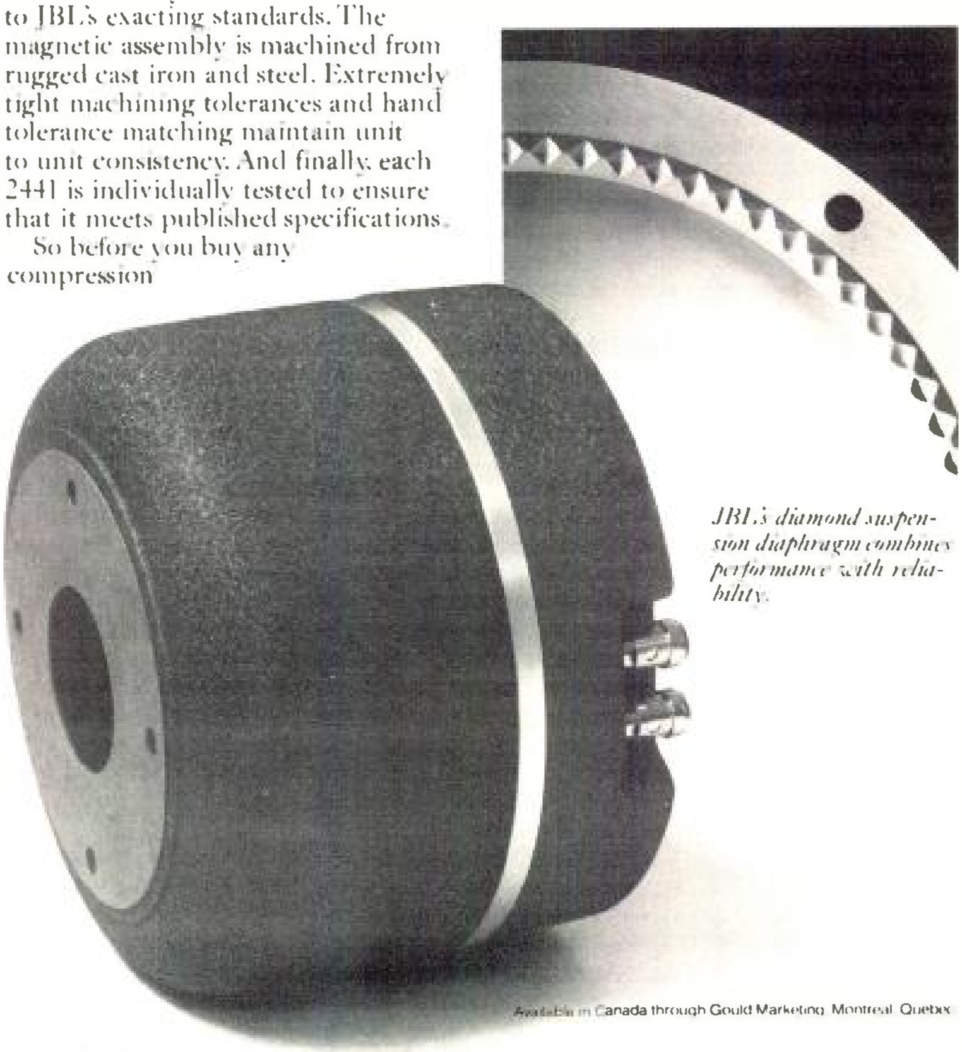
Specifications		
Horn Throat Diameter	50 mm	2 in
Nominal Impedance	16 Ω	
Power Capacity	70 W continuous program	
Sensitivity (1 watt, 1 meter)	111 dB SPL (on axis of a JBL 2350 90° radial horn)	
Frequency Range	500 Hz to 18 kHz	
Voice Coil Diameter	100 mm	4 in
Voice Coil Material	Edgewound aluminum ribbon	
Flux Density	1.8 T (18,000 gauss)	



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point, and most have two, at which clipping occurs. Beyond the clipping point, gain is zero; there is no transfer from input to output at all. Beyond this point, feedback isn't there any more. The amplifier quits working at that point on the waveform.

Now, our theorists have some fictitious "feedback factor" figure stuck in their head, and think it will do magic, even when gain disappears. Sorry to disillusion you: it won't. If you don't believe it—and I've encountered many who don't—go ahead and try it for yourself.

LOOK MA, MAGIC

Why is it that people get it into their heads that some figure like that possesses magical powers? A similar thing happens with statistical analysis: just feed your data into a computer programmed for statistical analysis, and hit the "standard deviation" button: *presto*, your "raw data" becomes more "reliable"! If somebody fed you wrong data, the standard deviation button won't correct that error, believe me. And if your feedback factor disappears when clipping occurs (as it does), nothing can bring it back—certainly not some magical number.

If that number really possessed that kind of magic, it would keep your amplifier working at full power, even when you unplugged it! I think you know better than that, unless there's some standby power provision.

But now we come to another fallacy that still makes the rounds from time to time. Feedback theory can be extended to loops within loops. And you can have positive, as well as negative feedback. Just enough positive feedback produces "infinite gain"—the thing goes into oscillation. Now if you have a small loop with positive feedback, encompassed by another loop with negative feedback, your "A" for the positive feedback loop can be raised to infinity, so you have infinite feedback!

POSITIVE AND NEGATIVE FEEDBACK

That should give us zero distortion, zero output impedance (or infinity, according to which way you go), and so forth. And the algebra seems to prove it. But now let's explore what we are really doing. Take the simple pieces first. When we apply negative feedback, the general trend is to reduce distortion—by the feedback factor at any given point in time or frequency—and to extend frequency response, subject to limitations imposed by phase shift, as shown by vector analysis.

Positive feedback does the reverse of negative feedback. So, it increases whatever distortion there is in the positive loop and it narrows the frequency response, with the usual result that when 100 percent positive feedback is reached, it happens at just one frequency, at which the equipment goes into oscillation. If you get more than 100 percent positive feedback, it may oscillate at multiple frequencies, or with a highly-distorted waveform.

So the concept of an amplifier with wide-band, 100 percent positive feedback, or even of a single stage within an amplifier with such a characteristic, is an impossible one, in the nature of things. But that doesn't show in the algebra. You write it down as a formula, so in theory such a thing exists.

If that single stage possesses distortion—and it could not have been perfect before the positive feedback was applied—then positive feedback, even at frequencies where it is not 100 percent, will increase the distortion the amplifier originally possessed by the same factor (the feedback factor) as it increases gain. Actually, when you reach 100 percent positive feedback, that produces oscillation at the frequency where it occurs. That oscillation, being unrelated to external input, is 100 percent distortion, although it would not normally be viewed that way.

Now, assuming you can control this by means of another feedback loop, outside this smaller loop—as can be done—the best that the external loop can do is to negate some of the damage done by the internal loop. If the internal stage had 1 percent distortion before positive feedback, and its gain was multiplied 99 times, it would have 99 percent distortion. (Perhaps that's oversimplified too, but it follows the simplified theory.) Now you plonk on negative feedback that brings gain down to half its value before the positive was added. The best that can do is to reduce overall distortion to one-half percent, and it might not do even that well.

How so? Well the algebraic distortion treatment uses a symbol for distortion, which just signifies what is presumed to be signal-related spurious signal generation. It could be second harmonic, third harmonic, or intermodulation distortion. It is generated by the signals being handled, but it is spurious in the sense that it appears in the output, not corresponding with an equivalent input. But a simple algebraic symbol does not indicate what, of the many possible varieties of distortion that can occur, this piece happens to be.

The algebra produces an expression that shows the input distortion offset by another distortion in the larger loop, so it comes out to zero. At least, that's what we once read in the literature, and it seems some are still "swallowing" it. If we have +d distortion, offset by −d distortion, the result is zero distortion. There are two things that can invalidate this.

First is the fact that +d may be, for example, second harmonic, while −d may be third harmonic. Then you don't have zero, but two forms of distortion instead of one. The other is that, even if the offset is correct, being of the same form, if you "add" −d distortion to +d (both being, say second harmonic), then you also have, which the simple algebra omitted to show you, d^2 distortion in the form of forth harmonic, although you may, in theory at least, have gotten rid of the second.

This could very well be the reason why lower orders of distribution became more noticeable after the advent of feedback. Many designers looked upon feedback as a panacea. So what did it matter if an amplifier had 10 percent distortion: just clean that up with lashings of negative feedback. To get the 10 percent second canceled would produce at least 1 percent of fourth harmonic that wasn't there at all before. And similarly with other forms of distortion.

Now, just assuming that 5 percent of second harmonic can be added to a pure tone without being noticed (actually it can't, but at one time we thought it could), if we use feedback to knock out 10 percent of second, and get 1 percent of fourth in exchange, that could very well be very audible. The level is only 40 dB below the fundamental, and it's two octaves higher. With other forms of distortion, the second (feedback) order distortion products would be even less related to the original signals from which they derived. ■

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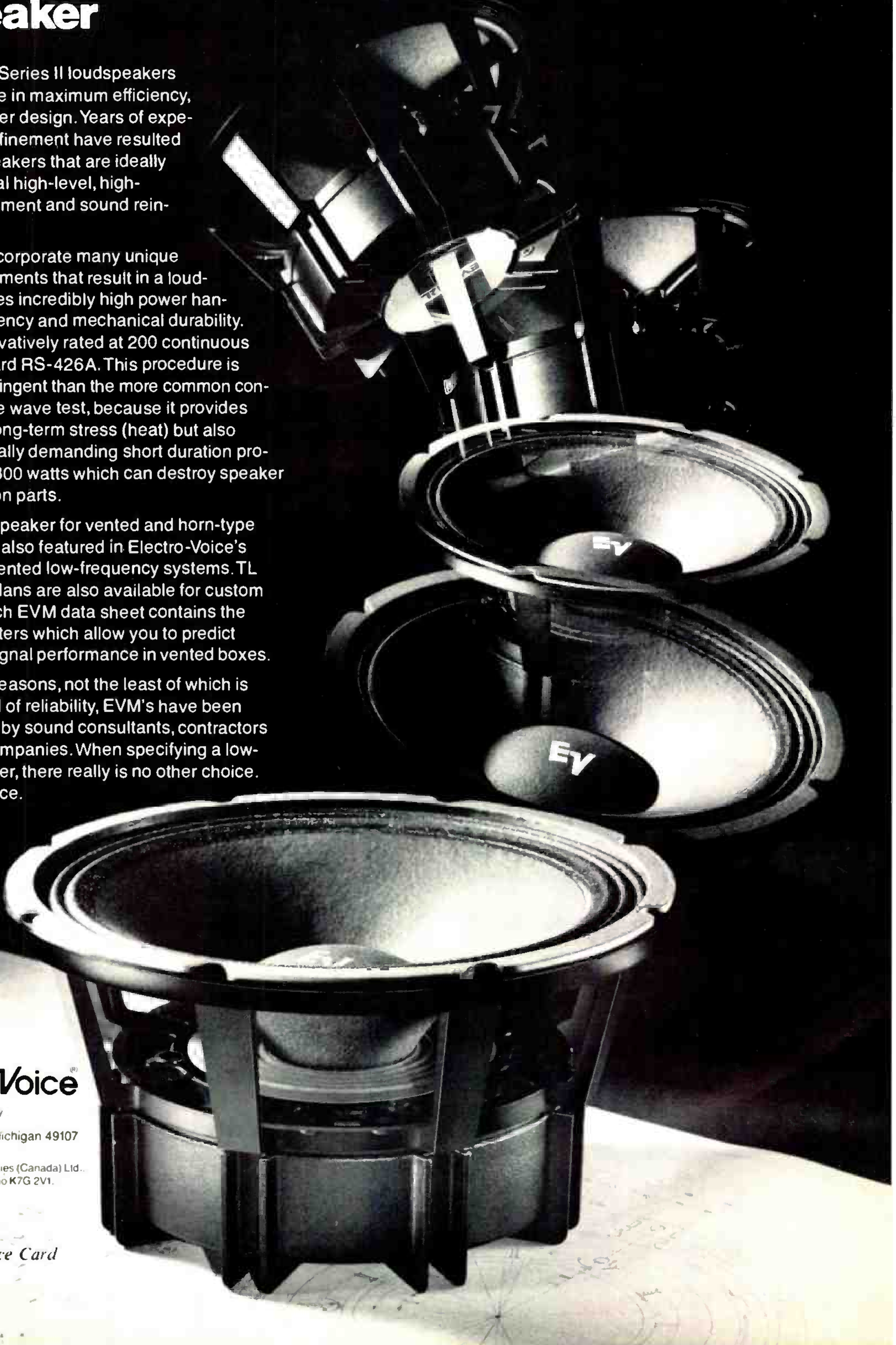

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Flipping and Flopping

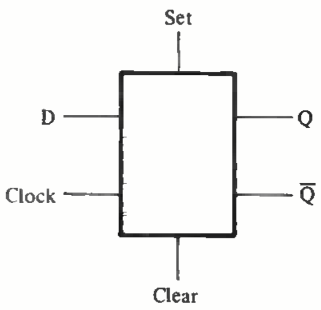


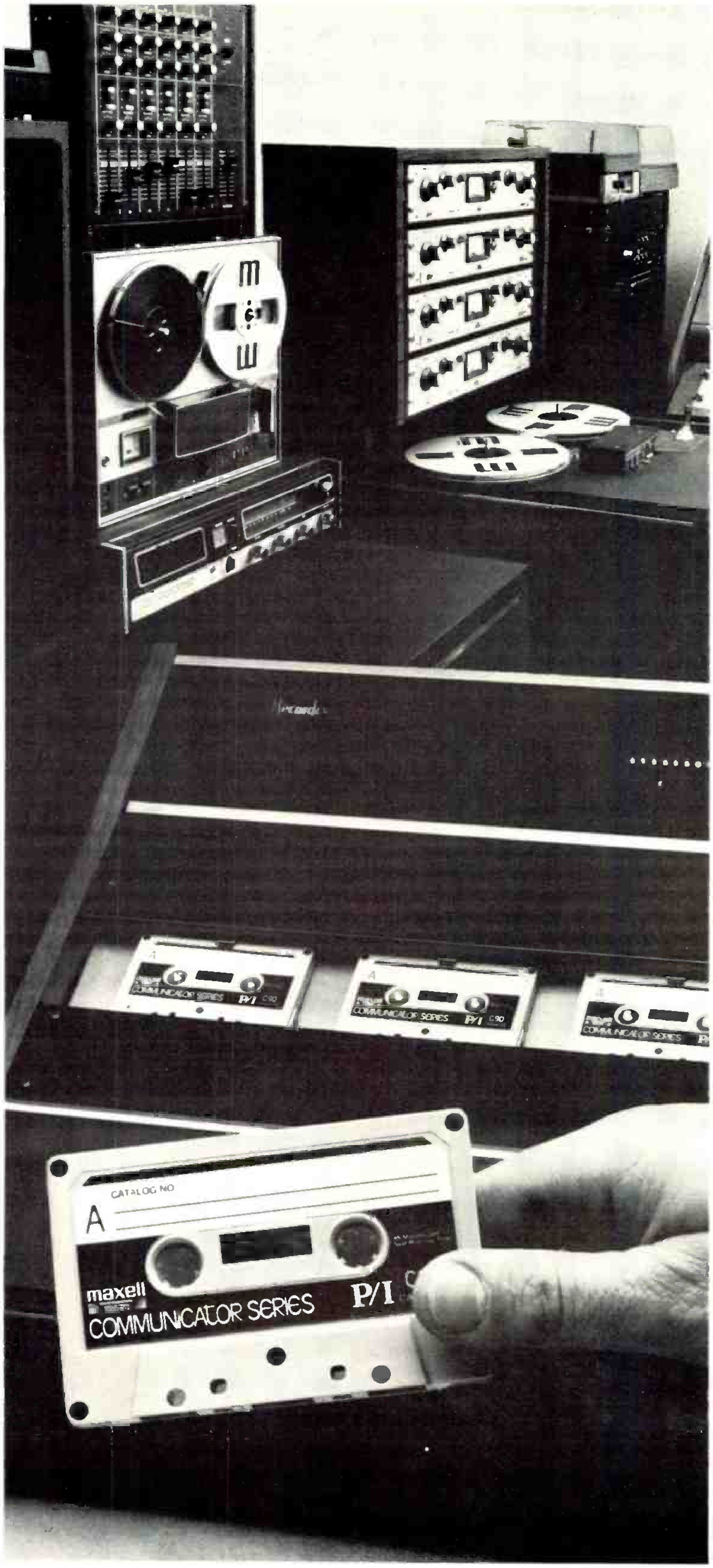
Figure 1. A typical clocked flip-flop.

• In our last article, we developed the concept of the clocked flip-flop based on a particular logical connection of ordinary gates. This discussion showed that the basis of clocking comes from the built-in "time-race" as the clock input makes a transition from low-to-high (positive clocking). We will now turn our attention to examining one flip-flop in detail by considering all of the specifications found in a data sheet. The 7474 will be our choice.

The operating characteristics of this device, shown in FIGURE 1, can be defined in the following terms: the value of the data on the D input will be transferred to the Q output when the clock input goes from low-to-high. Simultaneously, the Q output will contain the reverse information. The device also has two other inputs: Set (or Preset), and Clear. When *either* of these inputs goes to L (low), the flip-flop is forced to that state.

The first ambiguity in this flip-flop is that of defining the case when two of the inputs are active at the same time. For example, what happens if the Set input is active (low) while data is being clocked into the flip-flop? We cannot know the answer except by the manufacturer's definition. For the 7474 case, this is defined: the Set and Clear will override the clocking. Hence, if the Set is active, Q = H regardless of the clocking activity. No data can be entered because the Set input *overrides* the clock.

The second ambiguity is when both the Set and Clear inputs are active (low). The data sheet tells us that both of the outputs will be H, i.e. $Q = H$ *and* $\bar{Q} = H$. Although this appears counter-intuitive, the difficulty is really in our way of thinking. Q and $\bar{Q}$ are just outputs which can be "anything"; the names *suggest* that they are always complements, but the hardware characteristics are determined by logic—not by names. The difficulty in this case comes from the way in which we get out



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of the state where both Set and Clear are active. If the Set goes back to inactive before the Clear, the Clear will determine the stored state of the flip-flop, since it is still on. The reverse is also true. We should also point out that this definition is not universal. For example, the 74C74 (C-Mos version of the TTL 7474) has a different definition of active Set and Clear: both outputs are L, not H.

TIME SPECIFICATIONS

There are two classes of time specifications in a data sheet: the use specifications and the expected performance. For the use specifications we find such characteristics as maximum clocking rate, a characteristic of the device which must be respected if it is to work correctly. For expected performance, for example, we find such information as the range of delay time for the information to go from the D input to the Q output.

Let us now look at some of the more important use specifications. The flip-flop transfers the D information to the Q and $\bar{Q}$ outputs only when the clock goes from low-to-high; but it does not care about the D information before the clock or after the clock.

Set-up time is the amount of time before the clock that the D input must be stable in order for the Q output information to be considered reliable. This

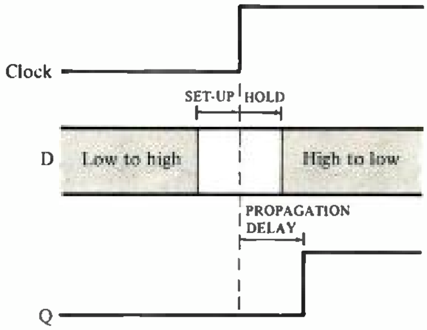

Figure 2. Set-up, hold and propagation delay times.

information must stay there, without changing, during the clock transition and for a certain amount of Hold time afterwards. After the Hold-time interval, the D information can again change without affecting the Q output. FIGURE 2 shows a graphical representation of this fact.

The D waveform can go to the H state at any time before the Set-up time and must then remain there until the Hold time has passed. It can then go low at any time.

The propagation delay is a performance specification since it tells us what is likely to happen. This specification has minimum, maximum, and typical delays. Worst-case design is usually concerned with the longest delay; however, in certain applications, the minimum must also be considered. Unfortunately, this can be a complex situation since we have temperature dependence, power supply dependence, load dependence, and the transition directions as variables. It is not always true that the time to go from high-to-low is the same as to go from low-to-high. Nor is it true that the Q will respond at exactly the same time as the $\bar{Q}$ output.

To make life still more interesting, the Set and Clear inputs may have different propagation delays than the clock. When timing issues are important, the designer also has the choice of using different types of parts with the same functional characteristics. The following devices have the same terminal relationships as the 7474 but different timing restrictions: 74L74 (lowpower TTL), 74H74 (high-power TTL), 74S74 (Schottky), 74LS74 (low-power Schottky), 74AS74 (advanced Schottky), 74ALS74 (advanced low-power Schottky), 74SC74 (enhanced CMOS). There is approximately a 100:1 difference in speeds between the 74L74 and the 74AS74. Since speed is often the most critical variable in a signal processing machine, there is a drift towards the 74S74 and the 74AS74.

Because a signal may effect a device in either of the H or L states, one often describes it as being either active or inactive. Hence, a Clear input is active when it clears the flip-flop. Active may correspond to H or L, depending on the device characterization. In our case, the Clear and Set are active low. New notation sometimes makes note of this fact. For example, the signal name would be Clear-L, thus telling us which state is the active one.

LOADING AND MARGINS

The output of a TTL device has a certain drive capability to feed its information to other outputs. Each additional input will load down an output such that, with enough loads, the signals will become unreliable. This characteristic is called Fan-Out (the ability to "fan out" the information to other inputs). In each logic family, there is a standard load for each input. For example, a normal TTL load is defined as −1.6 ma when the input is kept low. A high input produces some loading but this is mainly from leakage current. A standard TTL output is able to sink 16 ma when low. Hence, one standard TTL output can drive ten TTL inputs to the low state. The other logic families have different definitions of standard. A standard TTL-LS has a standard load of 0.4 ma and a standard drive of 8 ma for a fan out of 20. One can mix the families such that an LS output can drive four S inputs, or one S output can drive 50 LS inputs.

For certain TTL devices, the actual input to a circuit may be more than one standard load. The 7474 has a Clock and Clear input rated as two standard loads since it takes −3.2 ma to create a Low. Thus, a standard TTL output can only drive 5 7474 clock lines. Buffer-type outputs generally can drive twice as many loads as normal devices.

The loading characteristics are simple but we should understand the basis for them. As the output transistor tries to draw more current, its saturation voltage increases. With no loads, a TTL output will be about 0.1 volts in the low state. When the loading reaches its maximum specification at maximum temperature, the output can be 0.4 volts (TTL) or 0.5 volts (TTL-S). Since the Low level input voltage is specified at 0.7 volts for TTL-LS, there is very little margin for noise. The issue is the following. The voltage scale is divided into three regions: a well-defined Low, an ambiguous region, and a well-defined High. Below 0.7 volts, all logic will respond reliably as if the input were a logic low. Above 2.0 volts, the logic will respond reliably as if the input were a logic High. In between these, the logic may treat it as either. In practice, logic thresholds are usually on the order of 1.2 to 1.5 volts although the above are given as worst cases.

The second issue is that of system noise. We cannot expect all signals to be referenced to a perfect ground. Since ground noise adds to the signal, a voltage which should be a logic L can be made into a logic H if the noise-plus-signal exceeds the allowed limits. Thus, we speak of noise margin as the noise voltage which can change the logic value. A fully-loaded Schottky logic has only

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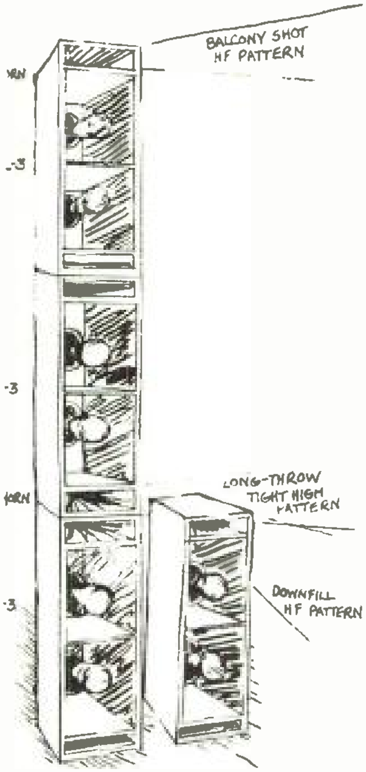


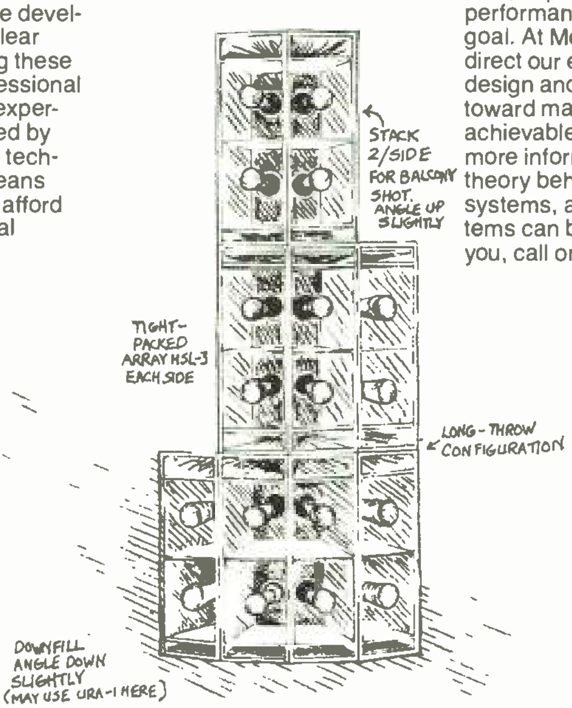
Every sound professional knows that loudspeaker arraying is a complex art. The history of sound reinforcement has seen a succession of theories and sophisticated calculations dealing with such concerns as coverage, acoustic power, and directivity. At each step in that succession, it has been the translation in the field of the theoretical ideal to the real that has pointed the way to improvements in the theory.

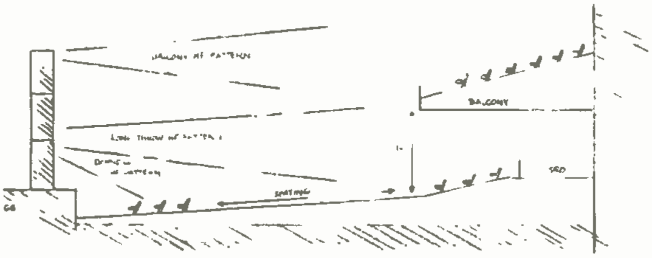
The Meyer Approach
At Meyer Sound, we've applied over a decade of research and field experience to the production of a growing line of reinforcement loudspeaker systems optimized for arraying, and we've developed sets of simple, clear guidelines for applying these systems. For the professional user, calculation and experimentation are replaced by a body of dependable techniques offering the means to make arrays which afford consistent, exceptional performance.

Polar Control
An important key to this performance is careful control of polar response. Meyer Sound reinforcement systems are designed to be coherent not only in terms of phase, but also in terms of propagation. For this reason, the crossover transition in Meyer systems is smooth and seamless, and frequency response remains consistent over long throws. In arrays, propagation coherence means smooth addition between adjacent units, minimizing lobing and producing a coherent image of the source behind the array. In practical terms, this means even, controlled coverage, greatly enhanced clarity, and little or no need for room EQ.

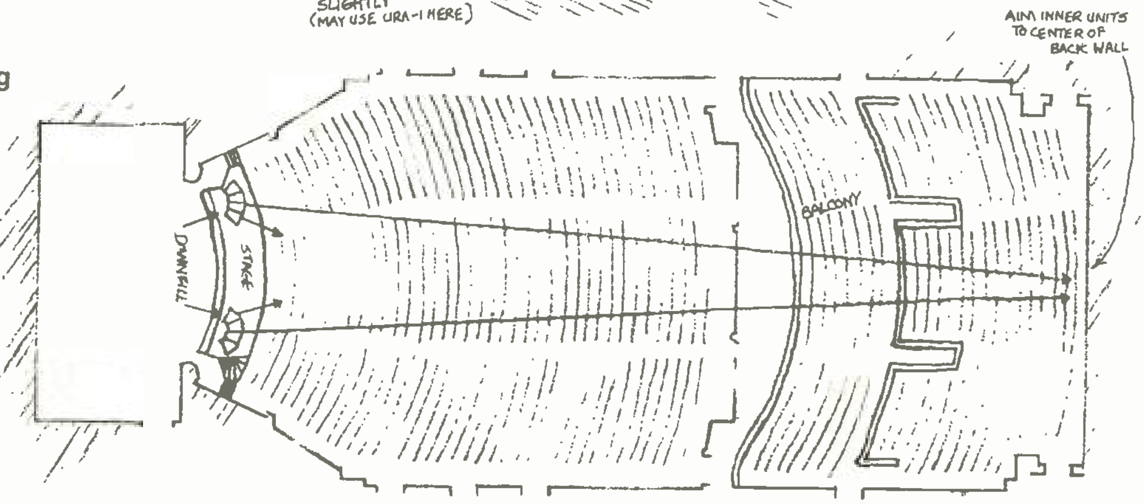
Modular Design
Meyer Sound reinforcement loudspeakers are designed as modular systems: full-range building blocks which offer the flexibility to meet a wide variety of demands. This means, for example, that the same product which serves for live music reinforcement in a 500-seat club can be used to make a large array for voice reinforcement in a 15,000-seat sports arena. Finally, since the array retains the performance of the modular unit with which it is made, its characteristics are predictable.

User Orientation
For the professional in the field, dependable real-world performance is the ultimate goal. At Meyer Sound, we direct our efforts in system design and documentation toward making that goal more achievable. If you would like more information on the theory behind our arrayable systems, and how these systems can be made to work for you, call or write us today.





Topic 1
Loudspeaker Arraying



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0.2 volts of margin in the worst case (about 0.5 volts in practice).

By not fully loading an output, the low level voltage becomes smaller and the noise margins become more. This is especially important with Schottky logic since the ground noise is large because of the high switching currents.

THE NOISE GLITCH

When we speak of noise in a digital system, we are not considering it in the same way as analog noise, because the origin is completely different. Analog noise comes from well-defined physical processes in the devices; digital noise comes from the way in which we build the system. A typical TTL output contains a circuit called a Totem-pole which is nothing more than an active pull down and an active pull-up as shown in FIGURE

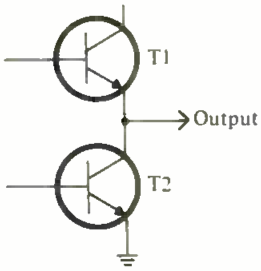


Figure 3. The totem-pole output of a typical TTL.

3. T1 is turned on when T2 is turned off and vice versa. In order to get the speeds high, the drives to T1 and T2 are either simultaneous or slightly overlapped in terms of current. This allows a high current to flow through T1 and T2 from power supply to ground. This current might be as high as 1 amp in certain devices, but the duration is only a few nanoseconds.

This current glitch can create a voltage glitch on the ground and power supply lines. Even with good capacitive bypass, layout and construction techniques determine the voltage glitch that results.

Other devices can see this glitch and respond to it. A TTL-S device will respond to a glitch that is only 2 nanoseconds long. The designer will thus see a 74S74 flip-flop "randomly" change state even when it appears that there was no input. Since the glitch may result from other processes unrelated to the 74S74, it is difficult to see it on an oscilloscope. Consider, for example, a counter which produces a glitch once every second. To find the glitch we would be looking for a 2 nanosecond pulse at a 1 second rate. This is a time ratio of about 1,000,000,000:1 which is not easy to see. Good instrumentation techniques by knowledgeable engineers are required to find such difficulties.

The issue of noise margin is extremely important. This requires good bypass capacitors and good ground distribution. At these frequencies, corresponding to microwave, not all capacitors are equal. Generally, ceramic capacitors are the only ones which will work and the loop area of the interconnection to the circuit determines the effective inductance. It would not be uncommon to get 2 nH of inductance for a physically large capacitor. A current of 1 amp through 2 nH for 2 nS is 1 volt! Most Schottky circuits are built on special wirewrap boards or, if PC, with multilayer ground planes. Ordinary two-sided PC board with Schottky logic will *never* work.

The situation is easier if low-power Schottky is used, since the current levels are lower and the noise margins higher. We can now see that the quality measure of logic is related to the current-speed product. One can always get higher speed with more current, but this tends to reduce the noise margins. Reducing the current solves the noise problem, but it also reduces the speed. Using this metric, we find that low-power Schottky has about the same speed as ordinary TTL, yet the currents are smaller. LS devices are thus preferred to ordinary TTL. Similarly, regular Schottky (S) has the same speeds as TTL-H (high power) but less current. Schottky logic will completely replace TTL and TTL-H logic. Similarly, the new advanced Schottky and advanced low-power Schottky will replace the current generation because of the improved speed-current product. ■

db Editorial

Working in the Sound Reinforcement industry does not always mean unloading tractor-trailers full of amps and speakers for a gig at Madison Square Garden. For like the recording industry, there's a lot more to sound reinforcement than freaking out with the stars at an endless succession of one-night stands.

Of course, there's probably more "glamour" in doing P.A. for the Stones than in reinforcing the Sunday sermon at the local church. However, when it comes to good old-fashioned "pro audio," could it be that there's more of it to be found in many churches than at many concerts?

With all due respect to Bob Ashley and friends, who have probably suffered through more Sunday morning audio penance than any collection of sinners really deserve (even consultants), a *good* church system could give the sound contractor a splendid chance to demonstrate his skill. For at the end of the sermon, the system is not dismantled and trucked off to the next night's stand. Instead, it is just turned off until next Sunday, when the cycle repeats. If the sound is sinful, the parishioners either fall asleep, or begin congregating elsewhere. In time, the sound contractor's reputation goes the way of all flesh.

By contrast, a lousy concert system is usually dismantled and safely out-of-town long before any sort of lasting impression sets in. Night after night, the road crew gets a chance to profit from yesterday's mistakes. Of course, if they don't get better after awhile, sooner or later they'll probably find themselves unemployed.

Meanwhile, back in church, if the audio committee is sharp, the sound crew may be asked to come back again (and possibly, again and again) until they get it right. And here's where a lot of folks get in trouble. The cure is often prescribed as "more of the same," when it should have been "pull the plug." However, pastors are just as human as other performers, and it takes a lot of faith to stand up without a trusty microphone in hand, and lots of watts nearby to help deliver the message.

The concept that a sound reinforcement system may be doing more harm than good is simply more than some performers (liturgical or otherwise) can handle. But when people walk away raving about the hardware rather than the performance (the software?), something is probably wrong somewhere.

Everyone (well, almost everyone) knows that a church system should not be conspicuous, and that the listener should only be aware of its contribution when it is turned *off*. Does the same thing hold true for the concert system? Sometimes.

Most of us would agree that a Pavarotti performance should not sound "canned." (If you don't know what a Pavarotti performance is, you've been locked in the control room too long.) On the other hand, a good sound system at a rock concert is often an integral part of the performance. Without it, the show would not—and could not—go on.

Unfortunately, there are simply too many performers who get done in by their on-stage audio hardware. Ironically, it's often the performer who demands the instruments of his own destruction. Fusion groups seem particularly prone to this kind of lunancy. It all starts out with the very necessary amp on the acoustic guitar. Then it spreads to the keyboards and through the entire rhythm section. Eventually, it terminates with six mics on the brass section.

It all began so that the guitar could be heard against the brass. But then the brass said "me too!", and again the guitar gets lost. So, bring on more power for the guitar. But then the brass says, "me too!" Repeat, *ad libitum*, and sometimes, *ad nauseum*.

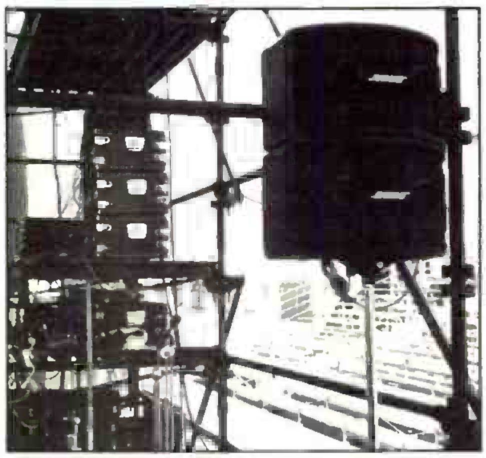
Some of us engineer-types often wonder if musicians listen to their own stuff anymore. Every now and then, we hear some amplified concert sound that would gross out a teeny-bopper. And yet, if we look, and listen (or, try to listen) beyond the electronics, there seems to be a sound source up there that's worth hearing. But you'd never know it by listening to what fills the auditorium.

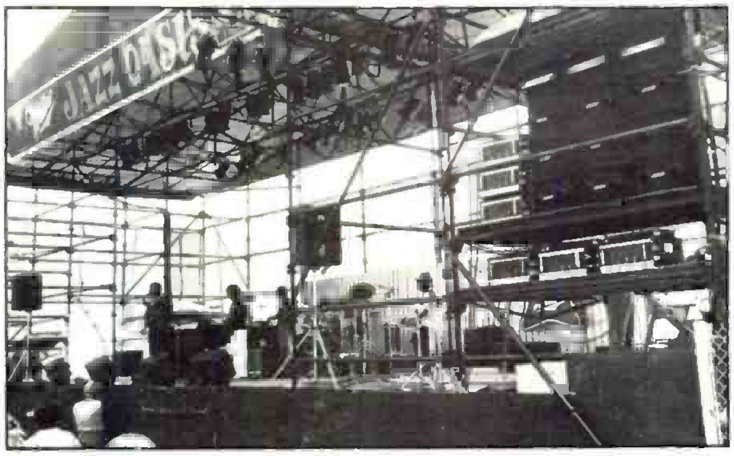
What went wrong? Ah yes, it's all the fault of "technology." You remember technology, don't you? It's that same evil force that forces rotten recordings on us, pressed into scratchy vinyl. It surfaces elsewhere as well—perhaps as a can of toxic waste that destroys a river or, worse yet, as "the computer" that destroys everything else.

Could it be there's a human-type or two at the bottom of all this? Why of course not! "The technology" does it all by itself.

Hard to believe, isn't it? Impossible to believe, we'd say. But there are some who really think this way. What do you think?

JMW





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An Overview of Sound Reinforcement

In the epic tradition of Homer and Tolstoy comes this look at sound reinforcement through the ages.

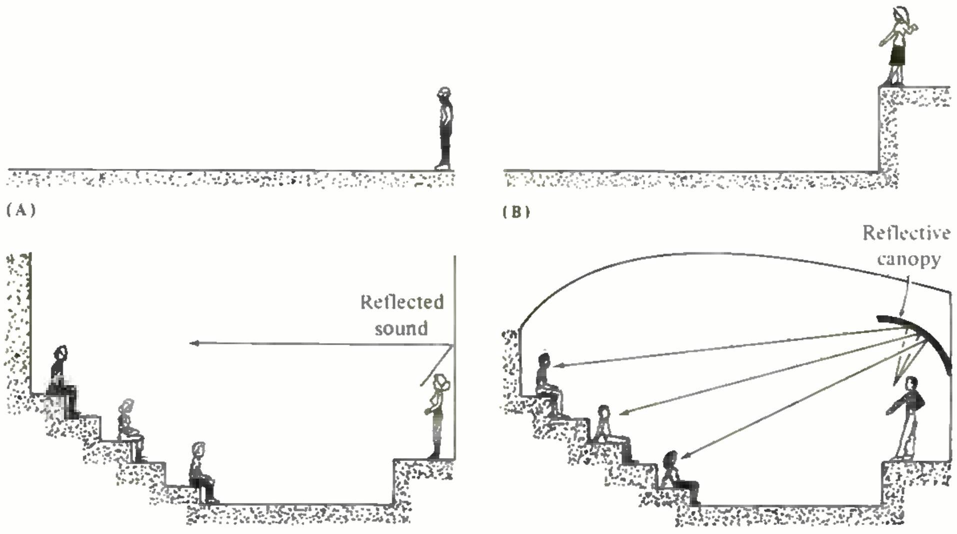

Figure 1. The evolution of natural speech reinforcement.

INTRODUCTION

THE BEGINNINGS OF SOUND reinforcement date back to antiquity. FIGURE 1 shows the evolution of natural speech reinforcement: early speakers addressing a crowd of people from a position at ground level observed that the sound of their voices was attenuated rapidly over the heads of the listeners (FIGURE 1A). Speaking from a higher position, as shown in FIGURE 1B, gave desirable eye contact and lessened the attenuation with distance. Succeeding steps resulted in the familiar open-air theater of the Greeks (FIGURE 1C), which made good use of reflected sound.

Later developments isolated the speaking-listening environment from the elements, and a final step added a reflective canopy overhead to increase the directivity of the talker toward the audience (FIGURE 1D). There remain today many medium-sized lecture rooms and small places of worship which, given good speakers, require no electroacoustical sound reinforcement at all. However, the general trend today toward multipurpose spaces has made some kind of electroacoustical reinforcement almost a necessity. The problem of addressing large crowds out-of-doors was the first to be met purely by electroacoustical methods. John Hilliard, in his historical review of horn loudspeakers (4), describes the use of a telephone receiver coupled to a typical phonograph horn as early as 1915. These were used in multiples to address an outdoor audience of some 50,000 people. More reminiscent of today's central clusters, a single grouping of "morning glory" horns was used in the San Francisco Civic Auditorium on Armistice Day, 1921. The advent of motion picture sound in the late twenties provided a sufficient base of hardware to take care of most indoor reinforcement requirements. Outdoor reinforcement, for the most part, remained with single-way paging horn devices used in multiples

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THE WESTERN ELECTRIC LEGACY

The Western Electric subsidiary of AT&T and RCA was responsible for most of the rapid development of motion picture sound during the late twenties and early thirties. In this relatively short time, many electrical and acoustical components became available for both large- and small-scale sound reinforcement applications. Many of these older components were in use up until just a few years ago, because the basic engineering and workmanship were of such high caliber. When the Audio Engineering Society first went to the Waldorf-Astoria Hotel in New York for its conventions in the early seventies, the facilities committee found that the old Western Electric loudspeakers used in the original distributed ceiling system, dating from 1931, were still in good working order! These loudspeakers were actually used for the first few years of AES conventions at that hotel.

One of the landmark components dating from the early thirties was the Western Electric Model 594 high-frequency compression driver. This device had a four-inch flat wire voice coil and exhibited conversion efficiencies on the order of 30 percent. A three-inch model was made by the Lansing Manufacturing Company prior to their merger with the Altec Service Company. It was not until 1954 that a permanent magnet version of the 594 was to be introduced: the JBL Model 375. Later versions of this driver included models made by JBL, Gauss and TAD, and have been given the benefit of extended frequency response and greater power handling capability.

In the years just before World War II, Western Electric was ordered by the FTC to dispose of its virtual monopoly in the motion picture theater loudspeaker and amplifier field. The divested group eventually became known as the Altec Service Company, and, as the war approached, a merger was made with the Lansing Manufacturing Company. Thus was Altec Lansing born. Five years later, Lansing left to form a new company, James B. Lansing Sound, Inc., and between the two of them Altec and JBL carried the Western Electric traditions forward.

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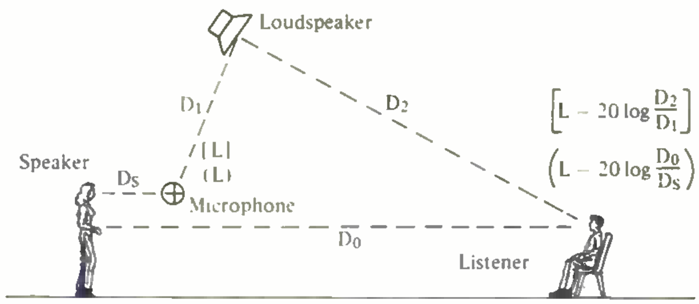

Figure 2. Details of an outdoor sound reinforcement system. Levels in parenthesis are with the system off; levels in brackets are with the system on.

Before he left Altec, Lansing and John Hilliard put the finishing touches on the A-4 "Voice of the Theater" system. This program gave rise to a family of low-frequency drivers and enclosures as well as high-frequency drivers and multi-cellular horns which, in one form or another, have dominated high-quality sound reinforcement up until recent times.

THE RISE OF SOUND CONTRACTING

The post-war era saw the extension of Altec's distribution pattern not only through traditional theater dealers but also through the Graybar Company to electrical contractors. In the late fifties, Altec abandoned the Graybar connection and began cultivating a network of sound contractors, individual entrepreneurs whose business was bidding on the sound portion of large construction jobs or dealing directly with the end user.

With their ambitious program of yearly clinics for their contractors, Altec was forming the basis for engineered sound system practice which we know today. One properly spoke of *sound reinforcement*, not *PA*. That latter term was restricted to re-entrant paging horns and their relatively simple applications. The paging horn tradition had been maintained during the thirties by Racon and was admirably carried forward in the post-war days by University and Electro-Voice.

While Altec was covering the sound contracting field, JBL was carving out a niche in the musical instrument (MI) area. Through most of the sixties, JBL's high-sensitivity four-inch voice coil transducers were modified for MI use and were distributed by Fender, the pioneering electric guitar company. This association gave JBL something of an advantage in the early days (1960s) of concert sound reinforcement. But we are getting ahead of ourselves.

RATIONAL DESIGN TECHNIQUES: THE RISE OF THE PROFESSIONAL CONSULTANT

As well-trained as many sound contractors are, the really difficult fixed-installation sound reinforcement jobs are best designed by professional consultants. A consultant should bring to a job not only a broad background in architectural acoustics and a thorough knowledge of hardware, but also a record of successful designs (along with some less successful ones from which he has learned what *not* to do). Consultancy in sound reinforcement expanded rapidly after World War II. It was often a sub-specialty of large firms, which addressed such problems as architectural acoustics, noise isolation, lighting, air conditioning, and other environmental aspects of large public places. Characteristic of good consultants from the beginning has been an appreciation of both science and art. They respected the points of view of musicians, and at the same time they were able to quantify many concepts which had been pretty much left to chance or intuition. One of the first of these was the measurement of speech intelligibility, or *articulation index*. Work in the late forties by Steinberg and French (6) and by Beranek (1) in the early fifties led to the quantification of articulation index and laid the groundwork for estimating performance in the area of speech intelligibility based on the parameters of a sound system and the characteristics of the hall in which it was located. Later work in this area has been carried out by Peutz and Klein (5).

During the sixties, Paul Boner provided a simple analysis of the gain of a sound reinforcement system (2) and pioneered the concept of narrow-band system equalization in order to minimize acoustical feedback. Because it is so instructive, we present in a modified form Boner's derivation of the equation which yields the maximum gain of a sound system. In FIGURE 2, we show a simple outdoor reinforcement system. The gain of a reinforcement system is defined as the *increase* in level perceived by a given listener when the system is turned on, as compared to the level he hears when the system is off. Let us assume that a talker produces some arbitrary level L dB at the microphone. With the system turned off, the level at the listener will be, by inverse square law attenuation:

$$L - 20\log(D_O/D_S).$$

Now, with the system turned on, the gain around the microphone-loudspeaker loop can be increased up to the point where unity gain exists; that is, where the loudspeaker produces a level at the microphone equal to that produced by the talker, L dB. If the loudspeaker produces a level of L at a distance of D_1, then the loudspeaker will produce a level at the listener of:

$$L - 20\log(D_2/D_1).$$

Since the system gain is defined as the difference between these levels, we have a Potential Acoustical Gain (PAG) of:

$$\begin{aligned} PAG &= L - 20\log(D_2/D_1) - [\,L - 20\log(D_O/D_S)] \\ &= 20\log(D_O/D_S) - 20\log(D_2/D_1) \\ &= 20\log D_O - 20\log D_S + 20\log D_1 - 20\log D_2. \end{aligned}$$

Of course, we cannot operate the system at the point of feedback, and it is customary to add some "safety factor" (typically, 6 dB). Thus, we modify the equation accordingly:

$$PAG = 20\log D_O - 20\log D_S + 20\log D_1 - 20\log D_2 - 6.$$

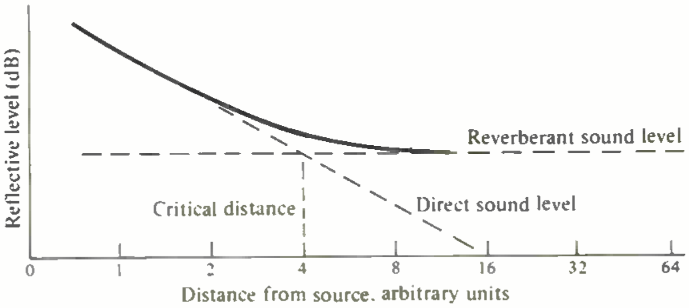

Figure 3. Details of an Indoor Sound Reinforcement System, showing the concept of Critical Distance, D_c.

$$D_c = 0.14\sqrt{QR},$$

where Q = directivity factor of the sound source, and
$R = S\bar{\alpha}/(1-\bar{\alpha})$
where S = total surface area in the enclosed space,
α = average absorption coefficient.

This equation is independent of units; as long as we consistently use meters, feet, or even inches, the answer will be the same. Note the inverse role of the microphone-to-talker distance, D_S. If that distance is halved, the potential gain increases 6 dB. There is a direct relationship with D_1; the farther the loudspeaker is from the microphone, the greater the gain.

For indoor systems, the analysis becomes somewhat more complex. However, there is a remarkable simplification which takes place when both the listener and microphone are in the reverberant field of the loudspeaker. In such an environment, as we move away from a loudspeaker or a talker, the attenuation of sound follows the inverse square law up to some point (depending on how directive the sound source is and how live the room is) where the direct sound field and reverberant sound field are equal (see FIGURE 3). This distance, measured from the

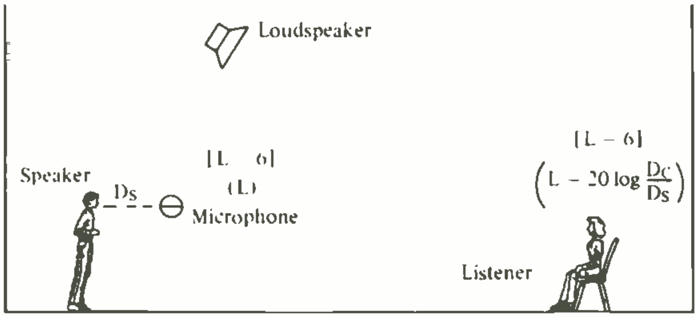

Figure 4. Levels in parenthesis are for system off; levels in brackets are for system on.

sound source, is called *critical distance*, D_C; as we move beyond D_C there is relatively little further attenuation of level. The potential gain equation under these circumstances reduces to:

$$PAG = 20\log D_C - 20\log D_S - 6.$$

Details of this are shown in FIGURE 4. Note that the only variable in this equation is D_S, the D_C term being more-or-less fixed by the environment.

Boner also determined the effect of additional microphones on system gain. Each time the number of open microphones is doubled, assuming they all contribute equally, the potential gain of the system is reduced by 3 dB, as shown in FIGURE 5.

Boner also pioneered the concept of narrow-band equalization to minimize feedback. Proper practice of narrow-band equalization has always been time consuming, and most successful examples are in large installations where microphone set-ups and gain settings are more-or-less fixed.

MANUFACTURER'S TRAINING PROGRAMS

As the groundwork was laid for rational system design, various manufacturers mounted their own training programs. Some manufacturers have taken a strong theoretical approach, attempting with some degree of success to teach contractors and dealers the rudiments of acoustical design. Other manufacturers have taken a more practical, equipment-oriented view of training and have stressed the application of their own gear. Unfortunately, no manufacturer has properly stressed the importance of good wiring and grounding practice and general shop competence. These are among the difficult lessons contractors must learn themselves.

Altec must take the credit for leading the way in contractor training. Their early clinic programs were expanded in the late sixties to include their "Acousta-Voicing" seminars. Subsequently, in the early seventies, JBL and Dukane set up training programs for their dealers and contractors. An important part of these programs are the loose-leaf training manuals prepared by the manufacturers. They have traditionally been given limited distribution, and they are not normally available outside the company's distribution channels. Consider yourself fortunate if you have been able to get your hands on any of them!

Number of Open Microphones	Effect on System Gain
1	0
2	-3.0 dB
3	-4.8 dB
4	-6.0 dB
5	-7.0 dB
6	-7.8 dB
7	-8.5 dB
8	-9.0 dB
9	-9.5 dB
10	-10.0 dB

Figure 5. System gain reduction due to open microphone.

While on the subject of training programs, we should mention the on-going sound system design seminars presented by Syn-Aud-Con. These programs have been held in various cities across the country since about 1973 and have introduced thousands of participants to the intricacies of sound system design and measurement.

SOUND CONTRACTING OVERSEAS

Sound contracting as we know it in this country hardly exists overseas. In most European countries, manufacturers themselves take over the role of sound contracting. Philips of Holland is perhaps the most visible, along with Siemens in Germany and Schlumberger in France. These large firms have the advantages of stability, broad research activities and financial leverage that no independent contractor could ever hope to offer. These companies more often than not provide consulting services as well.

Figure 6. A large outdoor rock concert. Such applications may be on the wane as music reinforcement turns to smaller venues with a greater accent on quality.

In Japan, the prestigous Sansei Engineering Company, once a private sound contracting firm, is now owned by Yamaha. Whether or not this signals the end of their traditional use of JBL and Altec components remains to be seen.

THE EMERGENCE OF CONCERT SOUND REINFORCEMENT

Beginning with the rock movement in the mid-sixties, many companies were organized to address the problems of touring rock acts. These "musical sound contractors" specialize in high-level sound reinforcement, addressing a market which is almost totally alien to the traditional sound contractor. The bigger of these firms are thoroughly professional in their work, and their interaction with certain manufacturers has had a beneficial influence on product design in recent years. These benefits have been mainly in the areas of power handling capability of existing transducers as well as the development of transducers for new and specialized requirements. As examples of this, we can cite the "ruggedization" of low-frequency transducers through the use of stiffer voice coil formers and higher temperature adhesives, along with the development of new eight-inch cone transducers designed to work in arrays or, as mid-range drivers, to cover the spectrum from 200 to 2000 Hz.

While masive outdoor festivals such as Woodstock and CalJam (see FIGURE 6) may be on the wane, music reinforcement is turning to smaller venues with a greater emphasis on quality—lower distortion and wider bandwidth. The time was when the low-frequency section of a large reinforcement set-up consisted of many reflex-type, horn-loaded enclosures stacked together. While such an approach as this produces lots of "woof," it does not usually go low enough to satisfy today's musical demands. Several manufacturers have designed high-powered 18-inch transducers to function as sub-woofers down to the 25 Hz range. The effect of these, in- or out-of-doors, is awesome, and FIGURE 7 gives some idea of conversion efficiencies and the resulting sound pressure levels when they are used in multiples.

For music reinforcement systems to exhibit flat power bandwidth out to 15 or 18 kHz, some kind of ultra-high-frequency device must be used in multiples. In just the last couple of years, many manufacturers have introduced "ring radiators." These are compression devices designed to cover the octave-and-two-thirds above 7 kHz with high output levels. For years, this product area had virtually been monopolized by one manufacturer.

Another movement in music reinforcement involves the use of multiple arrays of direct radiators covering all but the very highest part of the spectrum. This began with the "Grateful Dead" back in the early seventies, but progress has been made slowly. The advantage of this approach is the reduction of the inevitable distortion which accompanies *all* high-frequency systems when they are driven at high levels. The disadvantages have to do with size, cost and power requirements. The return to smaller venues may help things.

at mid-band. Thus, the system's power bandwidth, the ability to deliver full output at the highest frequencies, will be limited at higher drive levels. Under these conditions, a multitude of small high-frequency ring radiators will be needed to restore the required power bandwidth. Individually, these devices may handle only about 20 watts, but their extremely high sensitivity in the 7-18 kHz range (typically, 105 dB, 1 watt at 1 meter) enable them to do the job. Because they operate at such short wave lengths, they can be arrayed to ensure proper coverage at fairly large distances.

5. *System Drive Considerations.* Traditionally, reinforcement systems have been two-way in concept, with a transition in the 500-800 Hz range. Subwoofers or "tweeters" may be added, depending on the power bandwidth requirement at the frequency extremes. Current good engineering practice dictates that each range of the system be separately amplified, both for ease of control and for minimum distortion. When attention is turned to high-level music reinforcement, the two-way concept presents problems of both low-frequency power response and high-frequency distortion in the transition range. A three-way concept using intermediate horns covering the 200-2000 Hz range with appropriate small cone drivers tailored to the specific purpose works much better. Developments in this area are well under way.

Very-Low Frequency Output Capability

Half-space Reference Efficiency (1)		Electrical Power In	Acoustical Power Out	SPL Outdoors at: 10M	30M	100M	Typical Reverberant Field Level (2)
1 Unit	2%	200W	4W	98dB	88dB	78dB	98.3dB
2 Units	4%	400W	16W	104dB	94dB	84dB	104.3dB
4 Units	8%	800W	64W	110dB	100dB	90dB	110.3dB
8 Units	16%	1600W	256W	116dB	106dB	96dB	116.3dB

NOTES:
(1) One unit consists of a single 18" LF driver (Ref. Eff. 2%) mounted in a 340 liter (12 ft^3) enclosure tuned to 25 Hz.
(2) Room constant taken as 2462m^2, typical of a medium to large auditorium.

Figure 7. Very-low-frequency output capability.

FUTURE DIRECTIONS

Perhaps the best way to point to the future of sound reinforcement is to appreciate where the art has been. Let us take each area of technology and examine its evolution briefly:

1. *Low-frequency Systems.* The theater-type enclosures of yesterday are probably on their way out. They are big, expensive and boomy. Unless that boominess is desired for some musical reason, properly-ported systems will always work better and exhibit smoother power response. We can be thankful to Thiele and Small (7) for showing us all how to design properly-ported systems quickly—as well as for showing manufacturers what specific types of transducers had to be built for those systems.

2. *Sub-Bass Systems.* The spectrum below 40 Hz will become more important as time goes on—if for no other purpose than special effects. Again, ported systems are the most direct route to large amounts of power in the nether-frequency range.

3. *High-Frequency Systems.* We can thank Keele, along with Henricksen and Ureda (3), for bringing to the industry constant coverage horns. These remarkable devices can be equalized to yield both flat axial response and flat power response. They are the obvious choice where precise pattern control is to be maintained. Except for very special applications, they have pretty much rendered lenses, multi-cellular and traditional radial horns obsolete.

4. *Very-High-Frequency Systems.* For music reinforcement out-of-doors or in large halls, the power output demands at high frequencies will probably exceed the equalized high frequency output of a typical 800-18,000 Hz constant coverage horn. A typical constant coverage horn can be equalized, given the proper driver, so that its power output is effectively constant over its operating range. However, because of the high-frequency boost required to accomplish this, the system's power handling ability at very high frequencies will be much less than at mid-band.

6. *Design Options.* Many sound system designers tend to think along narrow lines; they typically want to make the same few system components work in a variety of applications. For speech-only work, there are many ways to devise linear arrays, electrically and physically tapered, to give excellent pattern control over difficult seating areas. A horizontal linear array stretching from wall-to-wall, for example, will be mirrored acoustically in both walls, producing an effectively infinite line array, with its characteristic 3-dB fall-off with doubling of distance instead of the 6-dB fall-off characteristic of point sources. This interesting situation alters quite a few of our favorite design equations and articulation index calculations. We can all look forward to a good bit of experimenting in this area. ■

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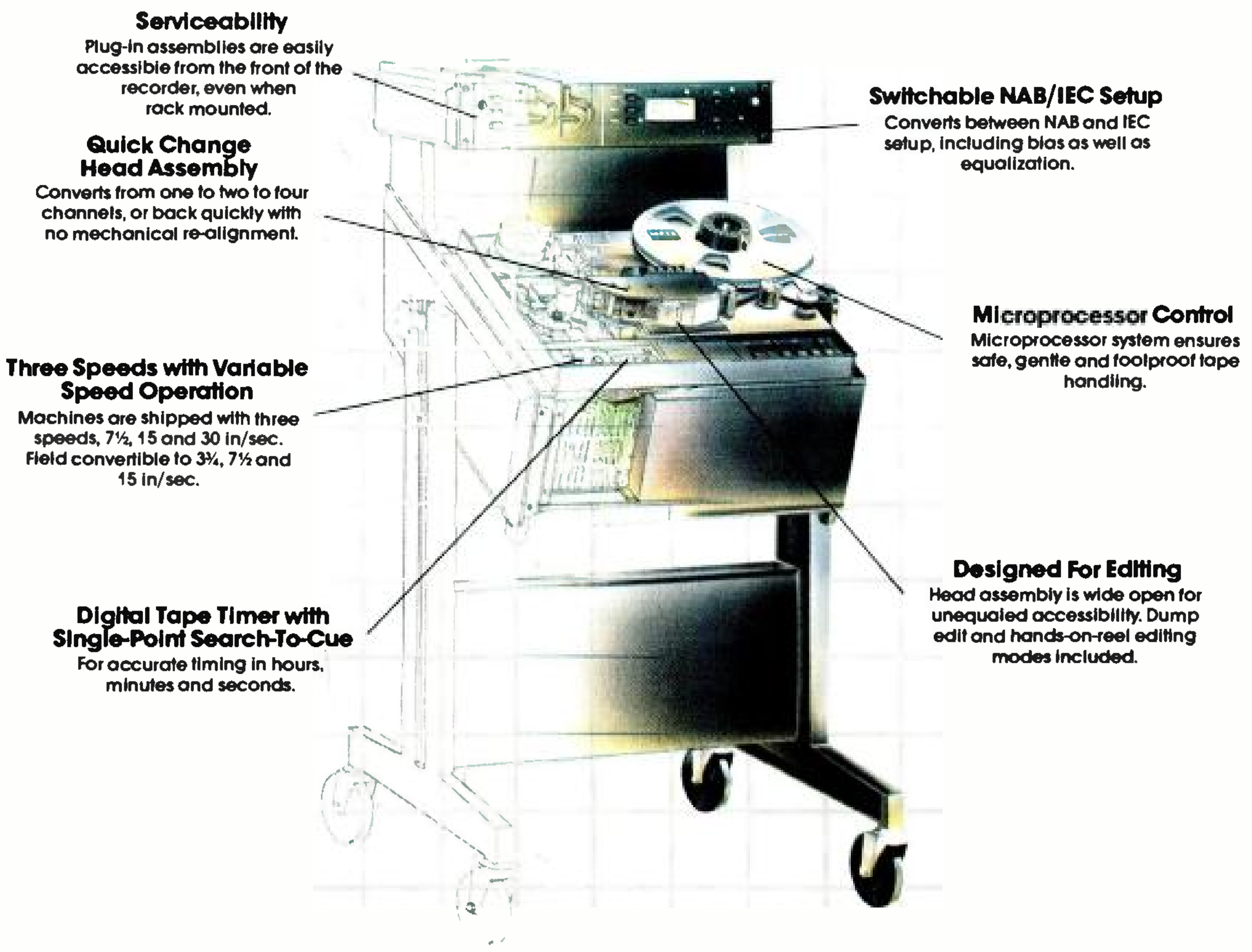

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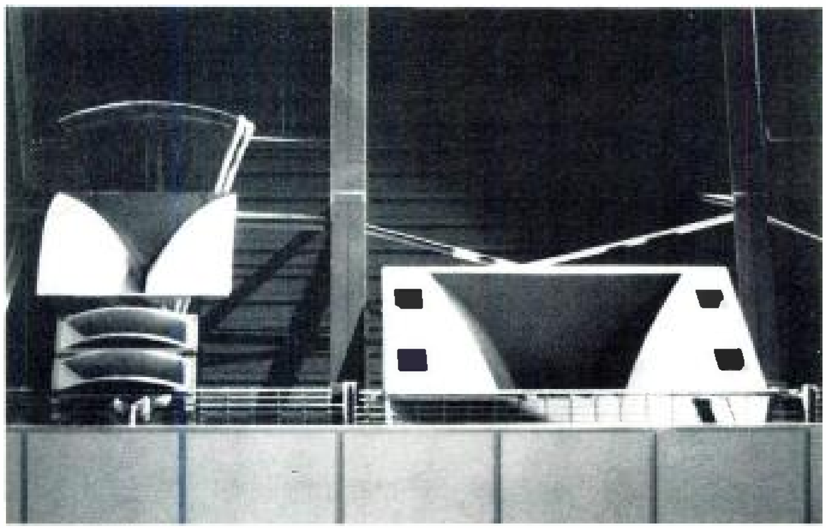


Close-up of central speaker cluster from below.

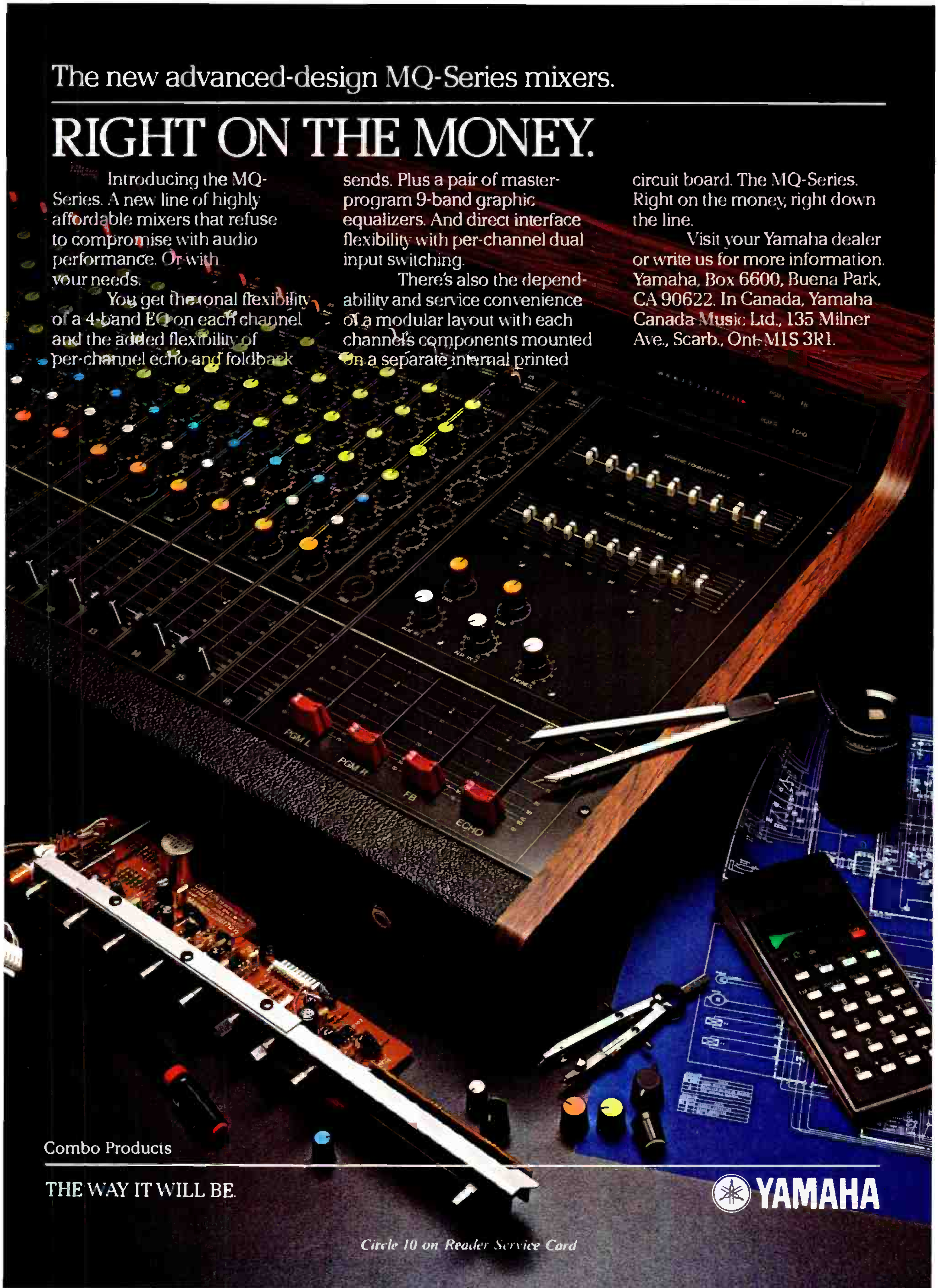
FEW CHURCH SOUND SYSTEMS have been so carefully planned as has been that of the new Willow Creek Community Church of Barrington Hills, Illinois. Ever since 1975, when the church was founded by a handful of hopeful Christian leaders, it had had no permanent home. The temporary systems set up in various locations had given Director George Everding and his associates a liberal practical education in sound reinforcement.

William Matthews is senior systems engineer at Ancha Electronics, Inc., Elk Grove Village, Ill.

Realizing the importance of church acoustics and electronic amplification, the building committee had solicitated advice from such knowledgeable sources as Lyle Yerges of Downer's Grove, Ill., George Augspurger of Los Angeles, Cal. and Robert Ancha whose company Ancha Electronics of Elk Grove Village, Ill. made the final design and installation.

The knowledge gained from these conferences justified the efforts, since the final results were all that the planners had desired.

Said Pastor William Hybels, "We have people at every service who tell us how marvelous they think the sound is—(they enjoy) the quality and the clarity. We are very well pleased."



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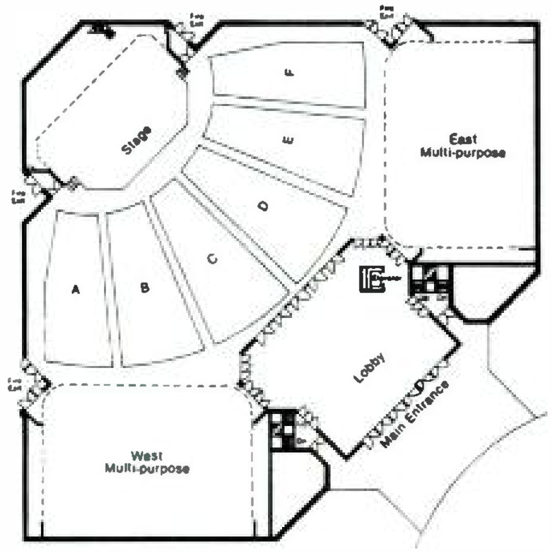
Lower Level

Floor plan of the lower level of Willow Creek Community Church.

Main Level



Floor plan of the main level of Willow Creek Community Church.

ACOUSTICAL CONSIDERATIONS

George Augspurger of Perception Incorporated made the acoustical treatment recommendations for the church. Total surface area of the sanctuary was 45,050 square feet, and the calculated average absorption coefficient at 125 hertz using the recommended acoustic treatment was 0.32. Using the Norris-Eyring formula then, the estimated reverberation time would be:

$$T = \frac{0.049V}{-S\ln(1-\bar{\alpha})} = 1.2 \text{ seconds, at 125 hertz,}$$

where V = Room volume (425,500 cubic feet),
S = Surface area (45,050 square feet),
$\bar{\alpha}$ = Absorption coefficient at 125 hertz (0.32).

Using the same formula for estimated absorption coefficients at one thousand and at four thousand hertz produced RT/60 figures of 0.65 and 0.68 seconds respectively.

The specification for background noise was an NC-25 and, with all air handling equipment in use, it would not exceed a noise coefficient of 30.

LOUDSPEAKER DESIGN

Due to the physical design of the Sanctuary, there was little anticipated early reflections to assist unamplified sound from the stage. Therefore, a very good loudspeaker system was a necessity to project clean sound to the audience. Three groups of loudspeakers were planned and installed on a gridwork above the stage. As specified, JBL components were used in the three identical short and mid-throw sections of the clusters. Each of these assemblies consisted of a 4550A low-frequency cabinet with two 2205H fifteen-inch loudspeakers; a 2360 bi-radial constant coverage horn with a 2441 driver mid-throw combination and two 2345/2440 horn-drivers for the front-throw. The two front horns were tightly coupled vertically to produce the wide horizontal and narrow vertical pattern required.

Above the center cluster, two vertically-stacked narrow angle 2356/2441 horn-driver assemblies were mounted and directed to cover the balcony. The resultant horizontal slot pattern was needed to prevent reflections from the rear walls.

Design criteria for each amplifier-loudspeaker circuit was to produce a sound pressure level of 105 dB to its assigned area.

Due to physical limitations it was not possible to vertically align the horns with the low-frequency loudspeakers. As has been noted in previous compromises, listeners did not detect any resultant problems in phasing or illusion.

AMPLIFIER CIRCUITRY

A biamplified combination of Yamaha and Ivie equipment was chosen. Each fifteen-inch low frequency loudspeaker was driven by one two-hundred watt section of a Yamaha P2200

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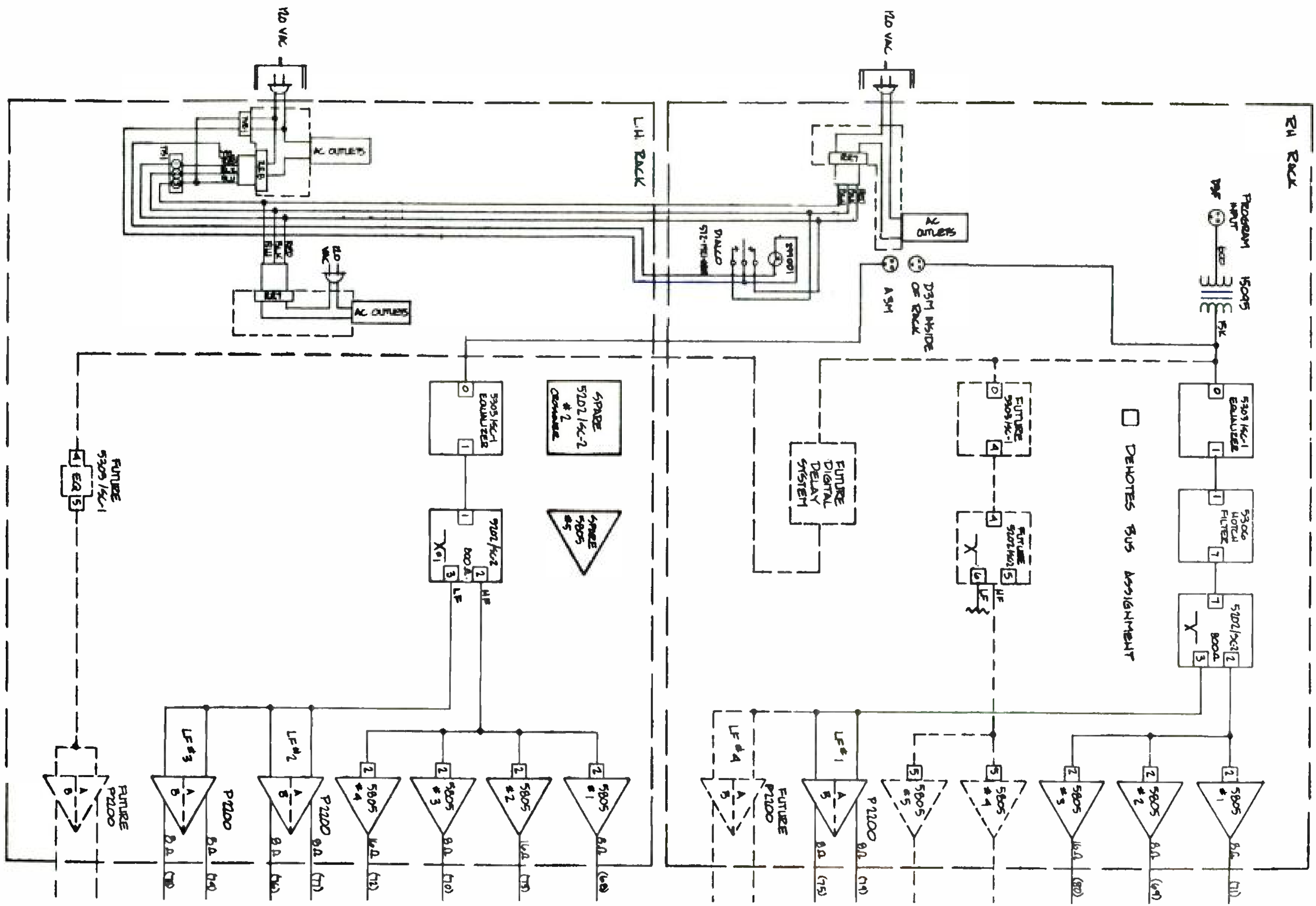
120 VAC
L.H. RACK
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AC OUTLETS
5305/ISC-1 EQUALIZER
SPARE 5202/ISC-2 #2 CROSSOVER
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DENOTES BUS ASSIGNMENT
5306 NOTCH FILTER
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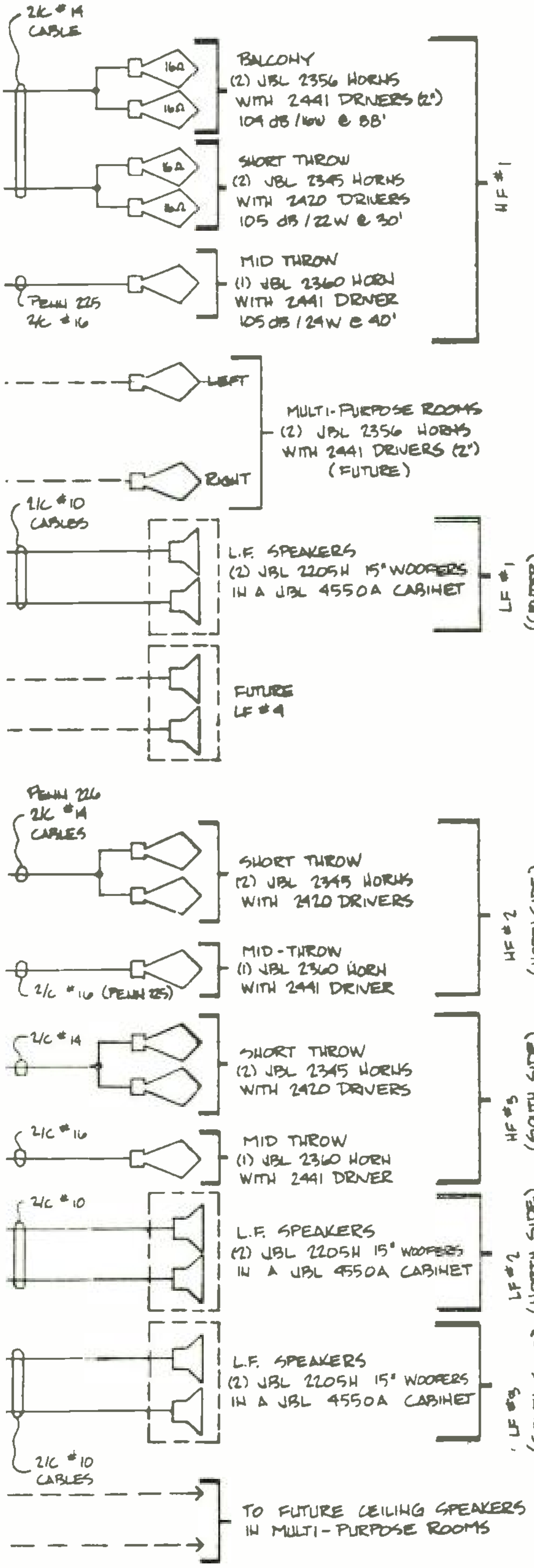


Block diagram of the Willow Creek sound system.

while each horn-driver circuit was powered by an Ivie 5805 one-hundred watt amplifier. An Ivie 5303 one-third octave equalizer and 5202 crossover-network was provided for the center circuits and another 5303 and 5202 was installed for the combined sides. A 5306 notch filter was added to the center design to cancel feedback modes from the microphones immediately below the center loudspeakers.

All amplifying equipment was assembled in two low-profile racks located in the control balcony. A Tapco thirty-two input mixer was installed nearby to drive the amplifying components. The control balcony is ideally situated in front of and slightly below the main balcony, directly in line with center stage.

ADJUSTMENTS AND EQUALIZING

Since it was essential that the near and mid-throw sections of all three clusters be identical, a thorough check of horn angles and placement was made prior to testing. Then pink noise was fed into the system with all controls flat. Equalizers, crossover networks and power amplifiers were adjusted for adequate headroom when the mixer was driven to its maximum output.

An Ivie IE-30 audio real-time spectrum analyzer was used at mid-point center seating to balance the audio spectrum above and below crossover. This readout was recorded in the Ivie's memory and used to balance loudspeaker reproduction in one of the side clusters to the same curve. Finally, the response of the other side cluster was adjusted to match.

At the same seating locations, the calibrated Ivie condenser microphone was placed and connected through a long cable to the analyzer at the equipment rack to facilitate adjustments to the equalizers.

The raw house curve was tailored to a plus or minus 1 dB response from 50 to 2500 hertz with a downward slope of 3 dB per octave to 12,500 hertz.

TESTS AND MEASUREMENTS

Ambient noise in the sanctuary was measured using the octave band mode of the real-time analyzer with all air handling equipment on and again with it off. A reading which equalled an NC-22 was achieved when quiet and an NC-33 was recorded with all blowers on.

Reverberation time was measured using the Ivie IE-30/IE-17 analyzer combination at the octave bands at which RT/60 had been estimated. Resultant readings of 1.5 seconds at 125 hertz, 0.6 seconds at 1000 hertz and 0.6 seconds at 4000 hertz compared favorably with the design criteria.

Loudspeaker coverage in the seating area was determined by reading the 4000-hertz octave band of the analyzer measuring pink noise through the loudspeakers. A respectable reading of plus or minus 2 dB was obtained.

The ultimate test was listening to varied amplified speech and several classifications of tape recordings at levels from eighty to one hundred decibels. Favorable opinions of the listeners confirmed the technical results.

Provisions for future expansion were made in the original design in the hope of church membership growth. Two multi-purpose rooms, one on each side of the balcony, were planned for possible conversion to additional audience areas. Utilization of all expansion spaces would transform the present capacity of sixteen hundred to well over five thousand.

Other features not normally found in a church are: A sixteen by forty-five foot projection screen which rises from the stage floor for motion picture and slide assists; a multi-scene programmable lighting console for custom stage illumination; a double-walled "floating" recording room, and two acoustically-isolated practice rooms.

Since the opening date of the church, the founding Christian leaders have been pleased by the better-than-anticipated attendance, particularly by young people. And, as one of them observed, "This makes it all worthwhile." ■

J. ROBERT ASHLEY

Audio Conversations—Church P.A. Systems, Part I

The following dialogue is a dramatic representation of the problems to be found in dealing with sound reinforcement for churches. Although the characters are fictitious, both the p.a. systems, and the problems discussed, are real.

INDIANA

JOHN HAS JUST returned from a consulting job in a small, rural Baptist Church. After hearing John describe the most effective cure for their audio problems, Bob has been laughing himself hoarse for several minutes.

Bob: I don't believe you cut the cord on their amplifier!

John: If you'll stop laughing, I'll tell you why. The church is 60 years old, small and with the organ, choir, baptistry and all the worship service up in the front. The ceiling is a steep gable with large wooden beams to support the roof. The side and rear walls are broken up enough that they do not serve as plane acoustical mirrors. The place doesn't need a p.a. system.

Bob: Agreed. But, I'll bet that the place had at least three speakers on each side wall and probably two or three more where the resident tinkerer had tried to cure an "I can't hear" complaint.

John: I don't take losing bets. Incidentally, all of the 8-inch speakers had 16-ounce magnets and the nicely finished closed boxes had at least 20 liters volume. The voice quality from any one speaker was surprisingly good. It was just the usual problem, the congregation could hear all of the speakers equally well. After five minutes, they were mentally sleeping through the sermons.

Bob: Still, weren't you a little irreverent for charging that minister $50 just to cut the power cord on his amplifier??

John: (Expletitive deleted) no. Joe Blank had already charged him a C-note for drawing up a set of plans to install a pair of columns and a third-octave equalizer. The construction bid was over 3 kilobucks. Besides, it took me an hour to explain why he did not need a p. a. system. Considering my travel time, I didn't make much on that job.

Bob: I have never been able to make enough to pay for my

J. Robert Ashley is an engineering staff consultant for Sperry Gyroscope.

efforts on a church job. Most of the remuneration is a good feeling for helping where help is so badly needed. How did you convince him in just an hour that the church did not need a p. a. system?

John: I used the sound and eyes argument. I had him sit in the second pew and I went to the lectern mike and started reading from the Bible. I got up close to the mike to almost overdrive the amplifier and read very fast. Within two minutes his eyes were wandering and I knew he had lost track of my reading. After four minutes, I stopped, went back and switched off the amplifier, and then resumed reading in a normal speaking voice and at the same rapid rate. He stayed with me for nearly 10 minutes. I told him of the psychological conflict of the sound coming from behind and the eyes telling the listener that the talker is up front. I find it best to keep the explanation simple—don't snow them with two-bit words like inhibition or directivity index. Keep the conversation moving so that he won't ask about the situation in the rear of the church where the sound is equally fatiguing but the eyes and ears argument doesn't hold much water.

Bob: Did he ask the usual question about how to make the timid reader heard in the back of the church??

John: Naturally. I explained that their church is far enough from the road that traffic noise is not very loud. I asked if the timid readers were understood even using the p. a. system and his answer was no—he received lots of complaints about that. I suggested a training program to teach the timid ones the elements of public speaking. I've seen this work in other churches and I hope he has the patience and skill to sell the concept to his people.

Bob: Good work!! I've never seen a p. a. system help a timid talker and wondered how to correct the problem. Most tinkerers try turning up the gain, get a howl, make the timid one even quieter, etc. There's the great conventional wisdom that loud means intelligible—and it doesn't.

John: I'm going to go to their worship service next Sunday and brag about how well I heard the preacher from the back pew.

Bob: I'l. bet you a six pack of Coors that the resident tinkerer installs a new cord within six months—

MARYLAND

Our friends have just walked out of a completely filled 300-seat Protestant Church after a funeral service.

Bob: That was a thought-provoking and wonderful eulogy. Did you notice that the preacher had the attention span of everyone for the full service? Did you notice how well this crowd of strangers joined in for the hymns?? It certainly proves that churches are better off without p. a. systems.

Paul: Yes it does—but you and I are called old fogeys for suggesting that most churches don't need p. a. systems.

Bob: I saw the speakers scattered around—they look like vintage 1950 to me. When did they give up tinkering around with speakers and amplifiers?

Paul: Well, the history is fairly typical for older churches. When it was built in 1920, Rice & Kellog might have had a gleam in the eye, but their electrodynamic loudspeaker wasn't published until 1928. At first, only motion picture theaters could afford the expensive and bulky loudspeakers. It wasn't until after World War II that prices came down enough for churches to afford p. a. systems. In 1948, the old preacher passed away and a former chaplain was selected. He did a lot for the church and brought in a lot of GIs. One of them was a Navy sonar technician. With large crowds and some timid souls trying to lead the prayers, I suppose it was natural for someone to suggest putting in a p. a. system. Guess who was drafted to design and install the system?

Bob: That sonar technician!! Back in those days the amplifier must have had a pair of push-pull 6L6s with a rating of 50 watts.

Paul: Right. They got the power rating by reading the RCA

Tube Manual and never bothered to measure it. I checked a few and found they would heat a resistor at 25 watts. That was plenty because the front speakers were close enough to the microphones so that it would howl long before the amplifier would overload. They moved those front two a half-dozen times and then turned them off. Then the choir members behind the preacher started complaining about not hearing the preacher.

Bob: I'll bet a sound level meter would have indicated to within 3 dB of the reading in the back pew. I suppose we did not then know about Auditory Backward Inhibition (ABI) and how the ears are different than sound meters.

Paul: They didn't need your two-bit words. They moved the two front speakers up and over the choir, tapped them way down in power, and declared it the best system in town. That poor sonar man, bless him, attended both services Sunday-after-Sunday to ride gain on the four mikes used. He never let a howl get through and it was surprising how loud the timid talkers sounded.

Bob: Did they still get complaints that the timid talkers could not be heard?

Paul: Yes, but there wasn't much starch in the complaints because the sound did seem loud enough. Nowadays, we call this hearing without understanding. It's the plague I've seen in most churches with p. a. systems.

Bob: I suppose the system was kept until there was a change of minister.

Paul: Yes, our chaplain moved on to a large city up North. There were several visiting ministers before a middle-aged chap with a strong, resonant voice was selected. Would you believe that it wasn't very long before he asked me what to do about that p.a. system?

Bob: But you don't live far enough away to come jetting in with a slide projector as a true expert. How did you turn it off without hurting the feelings of that sonar technician?

Paul: It wasn't easy. We waited until this fine man took a three-week vacation. I was asked to ride gain for those three weeks. The first Sunday, I got there a half-hour early with my Hewlett Packard Model 200 Audio Oscillator. As soon as a few people started talking on the church steps, I piped that trusty 200 into a mike input and swept the frequency and gain knobs like I was tilting a pinball machine. The racket in the church was horrible and sure enough, I got an arc on a dusty 6L6 socket, a ball of fire and a blown fuse. That preacher kept a poker straight face when I apologized for his amplifier blowing a frammatidazzle. I gave the timid talker a pep talk on speaking up and then stood in the back pew with my hand cupped on my ear to goad her to talk up—and she did. The preacher used that great voice of his to best advantage and got more compliments on his sermon than ever before.

Bob: Pretty sneaky. How did you keep it turned off?

Paul: Well, that frammatidazzle had to be ordered from Chicago and of course they sent the wrong size the first time. By the time the sonar technician got back, I had taken the amplifier home to try to improvise a frammatidazzle. He thought I was half-cracked but when he heard that great resonant voice of our preacher, he suddenly realized he was tired of riding gain all Sunday morning. Also, I got to the song leader with honest words about how much better the congregation was singing without the p. a. system and she got on her high horse about not needing a microphone. I think most folks got the p. a. fever out of their blood and knew they were better off without one.

Bob: I see a larger and newer Catholic Church down the street. I'll guess it was built in the mid-fifties and had to have a p. a. system to keep up with this church.

Paul: Right you are, except that the original reason given was that any 600-seat church had to have a p. a. system.

Bob: Without walking inside, I'll bet I can tell you about the sound system.

Paul: Since all Catholic Churches are supposed to be alike, you should be able to.

Bob: This isn't a matter decreed by the Pope but I sadly admit most of the Catholic Churches do have lousy p. a. systems. When was that church built?

Paul: 1953.

Bob: That was before the Vatican II reforms, so it has a choir loft with an electronic organ in the back.

Paul: Wrong—it's a good pipe organ.

Bob: That means they really have trouble with congregational singing now. Probably the sound system was designed by the architect or the electrical lighting consultant. It started out with six or eight 12-inch speakers mounted down from the ceiling. When it howled badly on day number one, one of the electricians got up and disconnected the front two speakers. The inexpensive microphones must have been high impedance with some single contact cable connectors and phone plugs along the way. At least the wiring met safety codes. In the excitement of moving into the new church, the fact that the system actually turned on and that adequately loud sound came out of each speaker convinced them that they had a good p. a. system. After all, every other church has the same kind of p. a. system—it *has* to be right!!

Paul: I suppose that is the conventional wisdom. For the first five years, our Catholic friends were sweating out the payments and couldn't have done anything even if they recognized trouble.

Bob: Did it ever break down long enough for them to realize it wasn't needed?

Paul: No. A local dentist took physics in college and is pretty handy with a soldering iron. They may have missed an occasional mass but never a couple of weeks in a row.

Bob: When did the Bishop rotate the pastor?

Paul: That was about 1962. The new priest immediately complained about the tinny sound of the system and let it be known that the Protestants down the street had a better p. a. system. After working on his teeth, the dentist got some free advice from our sonar technician. The dentist got a couple of friends to share the expense and put in some good cardioid mikes. They rewired the mike circuits with low-impedance balanced lines with Cannon 3-connector plugs and jacks.

Bob: That should have taken care of the hums, buzzes and squawks. Did the pastor think it sounded better?

Paul: No. They could get a little more gain before feedback but it still had that "church p. a. system" quality about it. Since it is a bigger church, it just can't sound as good with multiple speakers as the Protestant church with the same layout.

Bob: The next act is to put in better speakers.

Paul: Right you are! The dentist got his buddies to kick in a kilobuck—a lot of money in 1963—so that they could buy $50 made-in-California speakers. You should have heard the oohs and aahs as they unpacked each one tenderly, admired the cast frames, the huge magnets, and those shiny aluminum dust caps. The el cheapos that came out looked pathetic by comparison—and a couple had dragging voice coils.

Bob: So, the sound was louder, the distortion gone, the voice response was smoother—and people still slept through the sermons.

Paul: That dentist had to fix my bridge around 1965. He told me all about the great rebuilding project. I asked him what the last Sunday's sermon was about and he couldn't tell me a thing. In fact, he confided to me that the new priest was boring and didn't prepare his sermons very well.

Bob: That is sad. The congregation sleeps through the sermon because of the lousy p. a. system and then blames the preacher. Did you give the dentist the word on turning off the p. a. system?

Paul: Of course—and he considered my free advice to be worth just what he had paid for it. After all, what does an electrical engineer know about the physics of sound?

Bob: I suppose they endured that system until the next rotation of pastors. How did they get along with the changes in the

Catholic liturgy that put more emphasis on congregational singing?

Paul: Not very well. I hear stories about the unpaid organists not having good rhythm, the song leaders singing flat, and that no one sings with them. The good news is Christmas and Easter. They put together a good-sized choir, rehearse for about a month, and just raise the roof with good singing. The best music in town is their midnight mass. The congregation really joins in for singing the Christmas hymns. Maybe that is because they are so familiar.

Bob: No, Paul. It is because the choir is in the back of the church and that dentist hasn't rigged a mike back there. The only Catholic churches where I have found good congregational singing are the ones without p. a. systems.

Paul: Come on Bob, you can't blame everything bad in the Catholic church on p. a. systems.

Bob: That isn't much of an exaggeration. Remember how music has evolved in the Catholic Church. Until the past decade, music was a dialogue between priest and choir or a monologue from the choir loft behind. The big pipe organ evolved to fill the large European cathedrals with sound—you have to hear Bach at Notre Dame to really understand this heritage. A good priest with a 10-man choir doing a Gregorian Chant will raise goose bumps even if you not understand Church Latin. Turn on a p. a. system and the whole thing goes sour—the larger the church, the more mischief from the p. a. system.

Paul: What is being done about this mischief?

Bob: Not much. In the older churches, the organs have been left in the choir loft and the song leader brought to the front of the church. In my experience the church where this works best is a 200-seat Mexican Architecture Church in rural Boulder County, Colorado. This is one where the tyrannical old pastor turned off the p. a. system. The cantors have college educations in vocal music. A big university nearby with a good college of music and a large pipe organ means that one can hire good organists. The old Allen Electronic Organ is up in the choir loft. Once the singing gets started, the organ loudness is increased to punch through the melody line. The source of synchronization for the congregation is the organ—the singing is so loud that you can't hear the song leader beyond the first couple of pews. You have to hear it to believe it.

Paul: Don't most Catholic Churches seat more than 500 people?

Bob: Probably yes. When the song leader gets on the distributed speaker p. a. system and the organ plays from the rear you will hear many musical mistakes. If the musicians are non-professional, they will get out of step with each other. The resident tinkerer will be asked for help and he will often install another speaker near the organ console. Now the organist can stay in time with the song leader. However, you and I know that sound travels 345 meters per second and takes about a tenth of a second to get from the organ to the song leader. He thinks the organ is late by about a sixteenth note. Listen—every time you'll hear the tempo slow down in the firt bar of music.

The people in the church are caught in the middle of this conflict and don't know whether to follow the organ or the song leader—so they meekly switch back and forth. Usually, less than half join in the singing and these do not sing with vigor. They are afraid of making a mistake. This kind of singing doesn't do much for the Catholic liturgy.

Paul: Why don't you Catholics just move the organ up front, where it is in most Protestant Churches?

Bob: I don't know. If it is an old church with a pipe organ, there are obvious construction difficulties to overcome. Moving an electron c organ is no big deal, but I haven't yet been able to talke a s ngle pastor or parish council into moving the organ up front. I simply cannot convince them that the cause of their congregational singing difficulty is the sixteenth note of sound transit t me between the front and rear of the church.

Paul: Can you quote any examples of better music with the organ up front?

Bob: Yes. Every Protestant church I have been in where the organ is up front and the p. a. system not used for singing has excellent congregational singing. There is a five-year-old Catholic church 100 miles south of here which has the organ up front with the speakers flanking the altar. They don't even have a song leader and their singing is great. Just like that church in Colorado, the organ puts out a loud and clear melody line which synchronizes the singing. This works with the organ to the rear also—song leaders are not very necessary for congregational singing.

Paul: That's heresy.

Bob: Have you ever heard good congregational singing in any church where the distributed speaker p. a. system is used?

Paul: No.

Bob: Let's get back to the evolution of the sound system in the Catholic Church down the street. Most of the churches I know have some kind of a change to the sound system every ten years or so, usually when new people become involved.

Paul: Well, that dentist did move to a mountain town in Colorado and a new pastor was appointed about 1972. It didn't take him long to figure out that most people were sleeping through his sermons and that the congregational singing was anemic. After a big discussion, they decided to bring in a big city sound contractor.

Bob: And he put twin columns in the front of the church.

Paul: Well, what else could he do? The eight speakers on the ceiling were of the best brand he sold, the mikes and wiring were put in right by the dentist, and he had to do something different to justify his fee. The only other kind of p. a. system you see very often in medium size churches has a pair of sound column speakers, one on each side of the sanctuary, and about midway between floor and ceiling.

Bob: It sounded different—did he get paid for the job?

Paul: There was some dissention from the first because the bass response was not as good as it used to be—those p. a. columns have little speakers in them that cut off below 300 Hz. The contractor argued that that was the price to pay for good directivity and that now the eyes and the ears agreed that the talker was in the front of the church. They paid but were never convinced that they got their moneys worth.

Bob: The congregational singing probably went from bad to worse with an eighth note between the song leader and the organist. What did the sound contractor do about that?

Paul: He tried to sell them an electronic organ for the front of the church. No sale.

Bob: That would have probably worked better than their old system. Did the new resident tinkerer put a speaker back up by the organ console?

Paul: Yes, and the singing got a little better. About a year later, the pastor asked me to drop by to give a listen. I really didn't quite understand why the place seemed to sound more like a gym than it did with the eight speakers on the ceiling. At first glance, the twin columns ought to do better than that.

Bob: Most of the trouble was from that speaker up by the organ console. But, there is a booby trap in the twin-column idea which has snared most of the people who have tried them. The usual installation of a column speaker is for the bottom to be about eight feet up from the floor. This gets the directivity pattern over the microphones and brings up the acoustic gain. The booby trap is the usual smooth wall at the rear of the church. If you are in the front third of the church, you will get two bursts of sound spaced less than 10 milliseconds apart from the front speakers. Then, maybe 200 milliseconds later you will get a tight cluster of six bursts of sound reflected from the side and rear walls. Even a bounce from the front of the church after almost 400 milliseconds is surprisingly loud. With the eight speakers, there were just as many bursts of sound but they were spread out in time. The effects on intelligibility are quite different. The distributed speakers cause listening fatigue throughout the church. The twin columns are very intelligible in the back quarter of the church and the intelligibility degrades to poor as you move to the front. This is ABI—hearing without understanding—at work.

Paul: Your guesses about the evolution of that Catholic sound system have been pretty good. They should be getting a new pastor one of these days—what will happen next?

Bob: Someone will sell them on hooking up the speakers on the ceiling again, putting in a new, high-power, solid-state amplifier, and a third octave equalizer.

Paul: (Feigning innocence) Won't that take care of the troubles?

Bob: (getting red in the face) Hell No! I've never seen a third-octave equalizer do any good in a church, high school or even a concert hall. They will make the sound level meter go 10 dB higher but do nothing for intelligibility. Intelligibility problems are solved with acoustics and loudspeaker array design. First, one must correct some well-known acoustical difficulties with diffusion and absorption. Then, exactly one correct speaker array in the right location will make the amplified sound crystal-clear. Anyone who installs a third-octave equalizer is just proving that he doesn't know much about psychoacoustics or speakers. All they are doing is ripping off a bigger sales commission.

Paul: Calm down, Bob, I don't want to go to another funeral. What would you advise this new pastor to do?

Bob: Take out the p. a. system. They don't need one.

Paul: I agree, but it won't happen. After they get the five kilobuck quote for the third-octave equalizer with all the bells and whistles, they're going to be around asking for more free advice. What can I tell them?

Bob: First, take your side cutters and a crowbar up to the choir loft and rip out that speaker near the organ console. Give it to the Salvation Army. Then, get a music dealer to demonstrate an electric piano for the front of the church. Make sure that the amplifier and speaker can really belt out the acoustic watts—don't improvise with a guitar speaker. This stuff is going to take the biggest bite out of the budget. If this piano seems distorted or anemic, try another dealer. The piano speaker will probably work best in the right front corner for a Catholic church. Run the gain on the song leader as low as possible and watch the congregational singing get as good as you Protestants do it.

Next, show the pastor the Klipsch ad which calls for putting Heresy® in your church. I would prefer a Benson electrically tapered linear array speaker but these are not commercially available. The omni-directional Heresy® has a lower f_3 of 60 Hz and over 1.5 percent efficiency. It will operate with negligible distortion as a voice speaker. The most important thing is to follow the Klipsch advice on where to hang this single speaker. It goes to the top of the church, right above where the center of the communion rail used to be. From this point, the transit time for amplified sound is within about 15 milliseconds of the transit time from any talker in the sanctuary. If the speaker is run just loud enough to get a good signal-to-air-conditioner noise level, most listeners will get a clear acoustic location of the sound as right at the talker's head—this is the precedence effect of Joseph Henry in action. The reason this works when the twin columns didn't is the higher location. Sound from the speaker that is mirrored off of the back wall is soaked up in the back pews.

If you have a spare kilobuck in your budget, get a four-input automatic mike mixer. This will give the well-known 6 dB higher acoustic gain compared to running the four mikes open all the time. A more important advantage is that the open extra mikes circulate the amplified sound; you can think of it as electronically increasing the reverberation time. This would help the organ but it will degrade speech intelligibility. The automatic mixer is one of the new gadgets that is really worth the money.

Paul: That free advice should be worth more than what they will pay for it.

We leave our friends until next month. ■

KENNETH M. BOURNE

Professional Wireless Microphones Simplify Sound System Design

Wireless mics have improved to the point where today their sound rivals that of the high-quality wired mics. In addition, they are versatile and easy to use.

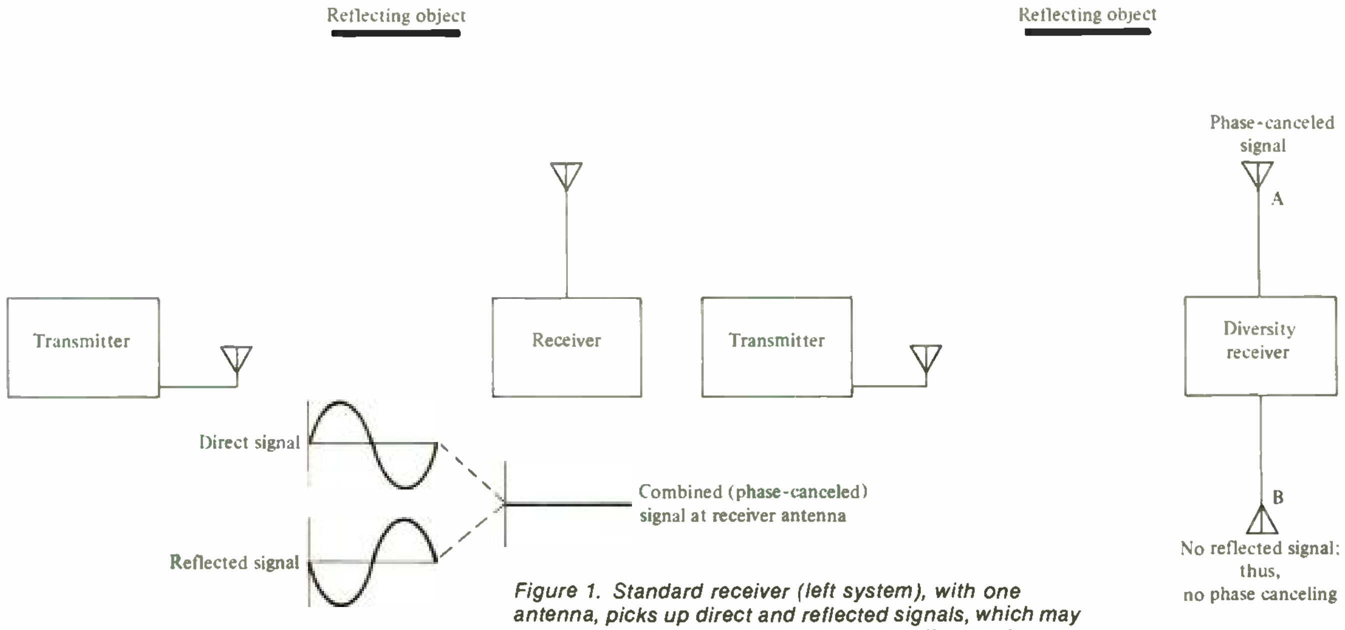


Figure 1. Standard receiver (left system), with one antenna, picks up direct and reflected signals, which may cancel each other (such as when two audio transformers cancel when connected together out of phase). Diversity receiver (right system), with two antennas, switches to receiver section (B) picking up strongest signal, thus avoiding dropouts caused by absorption or phase canceling on the other receiver section (A).

MANUFACTURERS OF WIRELESS MICROPHONES (sometimes referred to as "radio mics") began to appear in the early 1960s, to fulfill a growing requirement among performers for greater freedom of movement.

Some of the first wireless microphones operated in the 30-MHz band, using a hand-held transmitter with a telescoping antenna, or a lavalier unit with a wire antenna that ran from the RF output to around the user's neck. Later units used a trailing wire which was wrapped around the user's waist.

Early receivers were tunable, rather than crystal-controlled, and frequency drift and settability were occasional problems. Even when manufacturers introduced VHF high-band units, receivers were tunable, rather than crystal-controlled, as currently designed.

VHF HIGH BAND IS PREFERRED

Today, wireless microphone systems are available in several segments of the radio-frequency spectrum, with the following segments as the most common:

VHF Low Band	(25-50 and 72-76 MHz)
FM Broadcast Band	(88-108 MHz)
VHF High Band	(150-216 MHz)
UHF	(450-488 and 902-952 MHz)

Each of these segments has its particular advantages and disadvantages, but VHF high band (150-216 MHz) is the most favorable for most applications.

Several low-cost wireless microphones are now available on 49-MHz channels in the VHF low-band range. Some of these units even use the AM (rather than FM) mode, which limits audio frequency response and increases the possibility for noise interference. Even FM units on 49 MHz have poor signal-to-

Kenneth Bourne is the director of marketing for Cetec Vega.

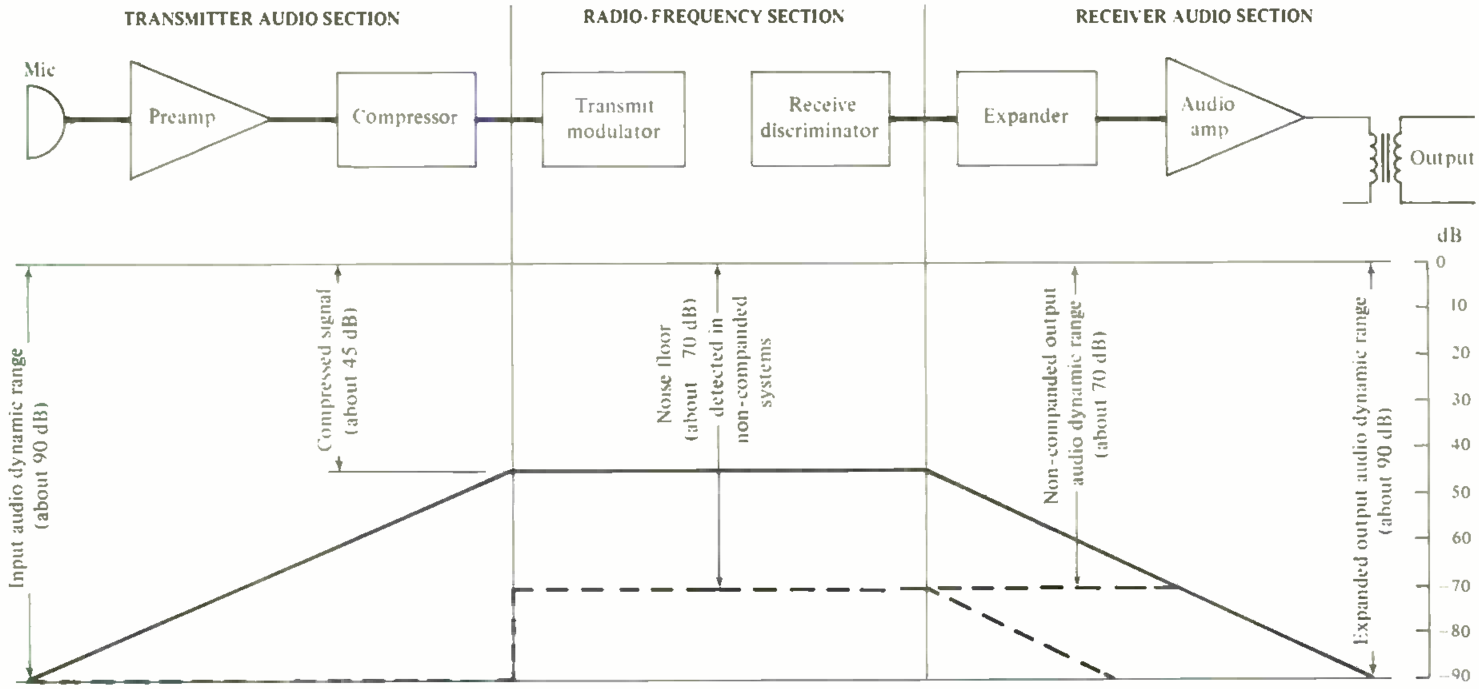


Figure 2. By compressing the signal to within the dynamic range handling capability of the transmitter/receiver RF section (45 dB versus 70 dB), and later expanding it, the original signal's dynamic range is preserved.

noise ratio. because of limited deviation in available transmitters.

Whether FM or AM. VHF low-band wireless microphones operate in the noisiest portion of the radio spectrum. and are most susceptible to interference. including long-distance signals propagated by ionospheric "skip" conditions. VHF low-band antennas must be quite long (almost 5 feet) to be efficient. and system range is limited. Furthermore. low-cost 49-MHz and other VHF low-band units typically are poorly constructed. and many rank in the toy category.

Wireless microphones operating between 88 and 108 MHz usually require the use of the customer's FM broadcast receiver. Typically. these units are also poorly constructed. and are often sold as toys.

VHF high-band wireless microphones are available from several manufacturers. and operate in a segment of the radio spectrum (150-216 MHz) that is very low in noise and interference. and that allows good range at low power (up to 1000 feet line-of-sight at 50 milliwatts RF power output). For broadcast and filmmaking applications. the Federal Communications Commission has allocated unused TV channels between 174 and 216 MHz (Channels 7-13) for wireless-microphone use.

UHF wireless microphones in the 450-MHz band are similar to. but not as common as. VHF high-band systems. and frequency assignments are limited. Range is not quite as good as at VHF high band. except for somewhat better building penetration.

UHF systems above 900 MHz typically are of good quality. but range is much more limited than at VHF high band. Signal dropout is common in many areas where VHF high-band signals are "full quieting." Manufacturing costs are much higher for 900-MHz equipment. and system prices therefore are higher than equivalent VHF high-band system prices.

Wireless microphones are even available above the radio-frequency spectrum. operating at infrared-light frequencies. Range is inherently quite limited.

WHAT ARE THE MOST IMPORTANT SPECIFICATIONS?

VHF high-band frequencies (150-216 MHz) offer the best range with least noise and interference for professional FM wireless microphone systems. However. many specifications and features must be considered carefully before selecting a system.

First. both the transmitter and receiver must be crystal-controlled. and must use high-quality oscillator components for a frequency stability rated at 0.005 percent.

From microphone input to receiver output. the overall audio

Coaxial Cable Length (in feet)	RG-58/U standard (dB loss)	RF Line Amp* Needed?	RG-8/U polyfoam (dB loss)	RF Line Amp* Needed?
50	3.0	No	1.0	No
75	4.5	Optional	1.5	No
100	6.0	Yes	2.0	No
125	7.5	Yes	2.5	No
150	9.0	Yes	3.0	Optional
175	RG-58/U Not Recommended		3.5	Optional
200			4.0	Yes
250			5.0	Yes
300			6.0	Yes
400			8.0	Yes

Figure 3. Calculated RF signal losses for VHF antenna cables, with recommendations on when to use an RF line amplifier. When using an RF line amplifier, it should be set for 1 dB more gain than the calculated cable loss. Alternately, with a weak signal, the wireless receiver's VU meter should be observed for no further increase in audio level while increasing the line-amplifier gain.

frequency response should be flat (±2 dB) from 40 Hz to 15 kHz, or ±1 dB from 100 Hz to 12 kHz. One of the highest quality systems on the market offers a compressor to maintain clean, unclipped sound with even the loudest noises, and total harmonic distortion of less than 1 percent (at up to 30 dB of compression). To preserve the dynamics of the original program, the signal-to-noise ratio should be better than 70 dB, or better than 90 dB in a companded system. (See the section on companding later in this article.)

Total harmonic distortion should be less than 1 percent. Emission usually is specified as 54F3, direct FM, ±12 kHz deviation, with 100-microsecond transmitter preemphasis and receiver deemphasis.

When used with a suitable antenna, the system range (from transmitter to receiver antenna) should be well over 50 feet (worst-case range due to obstructions or multipath reflections from air-conditioning ducts or other metallic objects), and even over 1000 feet in some line-of-sight conditions (such as in football stadiums).

DEFINING TRANSMITTER REQUIREMENTS

To assure uninterrupted operation, all transmitter connections should either lock or screw into place.

Professional wireless transmitters are available in pocket-sized or hand-held versions. Pocket-sized transmitters should be lightweight (about 5 ounces, including battery), and small enough to fit into a shirt pocket.

The input of a pocket-sized transmitter should accept any popular professional-type low-impedance (150-ohm) microphone, such as the Sony ECM-30 or Unex electret-condenser microphone. An external control should be available to adjust microphone input levels from about -65 dBm (0.22 mV) to -20 dBm (39 mV), to the threshold of compression. (The FCC

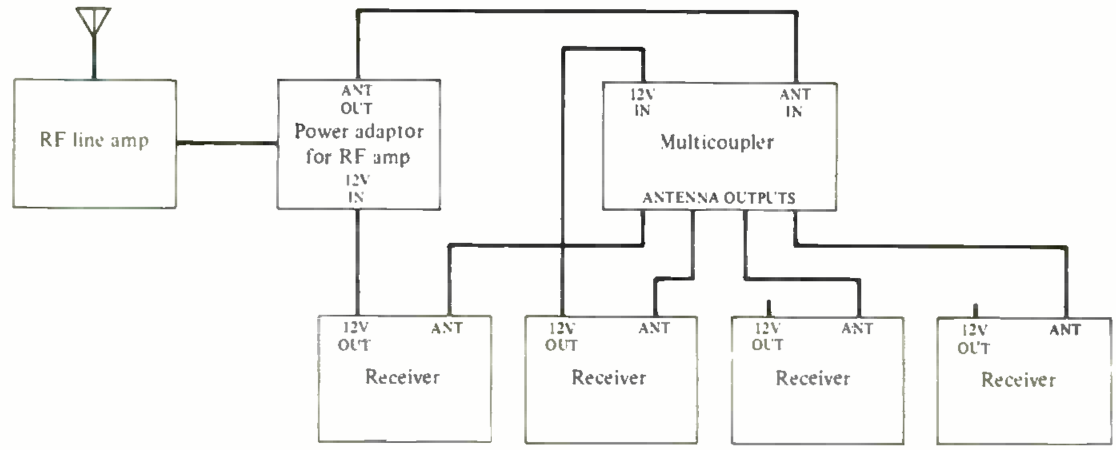


Figure 4. RF line amplifier with four-port multicoupler allows one antenna to be used with four receivers. (RF line amplifier is not necessary with short antenna cable run.)

requires compression or some form of modulation limiting.) Ideally, the modulation limiter should be a compressor (not a clipper) with 5-millisecond attack and 200-millisecond release times. Onset of compression should be externally metered and controllable with an external mic-level trimmer. Above threshold, a 30-dB increase of input signal should result in only a 3-dB rise in modulation (10:1 ratio). The mic connector often is recommended to be a Lemo "Quick-Loc," a favorite in the demanding film industry because of its positive-mating gold contacts, push-lock security, and rugged strain-relief cable fitting. When bias is required for an electret condenser microphone, it should be obtainable from spare pins in the Lemo connector, eliminating the need for bulky and unreliable external battery supplies.

The pocket-sized transmitter case should have no sharp corners, be highly impact resistant, and be wearable in a costume or taped to a performer's body with safety and comfort. The case should be sealed to keep out high humidity (and rainwater), and to protect the circuitry from corrosive body acids. A separate slide-open battery compartment should be provided to allow battery removal without exposure of other circuitry. The transmitter should be designed to accept a standard 9-volt alkaline battery. If several batteries are to be stored for long periods of time, it is recommended that they be stored in a cool location, such as a refrigerator, for extended shelf life. (Rechargeable NiCd batteries usually are not advisable, because of short operating time between charges, and because of heavier weight.) A transmitter with 50-milliwatt RF output should have approximately an 8-hour alkaline-battery life.

The antenna should be a quarter-wavelength flexible wire permanently attached to the pocket-sized transmitter for greater reliability and maximum output.

A professional hand-held transmitter should also be light-weight and as attractive as possible. Preferably, the antenna should be incorporated into the microphone housing, to eliminate unsightly dangling wires and "rubber duckies." If designed properly, the built-in antenna will radiate as well as or better than an external antenna. The case should have a contoured shape (to avoid the "flashlight" look) and should have a well-balanced feel.

The hand-held transmitter should be equipped with an industry-standard microphone capsule, such as an Electro-Voice EV-671 or Shure SM58 element.

Frequency response and other specifications should be the same as mentioned above for pocket-sized transmitters. An audio gain control should be provided on the bottom of the hand-held microphone case, to allow the user to adjust the mic's sensitivity. Ideally, optimum setup should be verifiable, such as with an adjacent LED compression indicator. The microphone should also include a power on/off switch plus a separate audio on/off switch to keep the receiver quiet (with a continuously transmitted signal) while temporarily silencing the mic's audio.

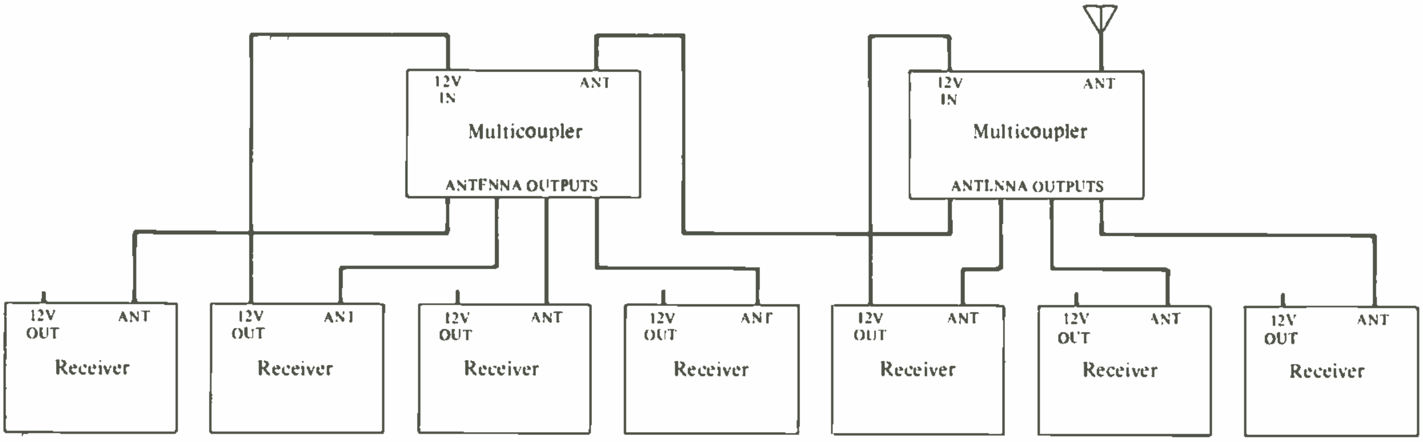


Figure 5. Two multicouplers can be chained to allow seven receivers to operate with one antenna.

The transmitter should be the type accepted under Part 90 of the FCC Rules and Regulations, if it is to be used in business applications, or under Part 74, if it is to be used in broadcast or filmmaking applications.

DEFINING RECEIVER REQUIREMENTS

The ideal professional wireless receiver is extremely sensitive, highly selective, and inherently stable. RF sensitivity should be less than 0.7 microvolt for 20-dB quieting. Ultimate quieting should be better than 70 dB with 1-mV RF signal input, and at least 70 dB squelch quieting (no signal input). Image and spurious rejection should be at least −70 dB. The design should include a true helical-resonator preselector and crystal-controlled local oscillator stage.

Full monitoring facilities should be provided to indicate system status. A professional wireless receiver should include a VU meter to indicate audio-output level. Metering should also be provided for relative RF signal strength and primary power (AC line or battery level). LEDs should be provided on AC-powered receivers (not battery-powered, because of current drain) to indicate when the power is on, and when an RF carrier is being received.

In a diversity receiver, LEDs should be provided to indicate which receiver section is selected. Diversity reception increases signal pickup reliability, with minimal dropouts or fades (noise ups) as performers move around a stage or set. (See FIGURE 1.)

Diversity reception is especially effective in eliminating multipath cancellation, in which a reflected signal arrives at the receiving antenna slightly later than the direct signal. Reflecting objects include air-conditioning ducts, chicken wire, metal benches, screens, etc. The reflected signal is out of phase with the direct signal, and the two signals will cancel each other, either partially or completely. This problem can occur even at short distances. However, when two antennas are used (each connected to a separate receiver section in a diversity receiver), multipath cancellation rarely occurs on both antennas simultaneously, and the receiver section with the strongest input signal is selected automatically.

Path loss and absorption are other types of signal cancellation that may be overcome by using a diversity receiver. As the transmitter user moves so far away from one antenna that normally a signal fade would result, he may still be within range of the other antenna, and the receiver section to which that antenna is connected will then be selected automatically. The same effect would occur if an interfering object were between the transmitter and one of the antennas.

The ideal diversity receiver contains two receiver sections in one package, with two antenna inputs, one for each receiver section. (A diversity receiver is superior to a single receiver with a diversity antenna.) The receiver section must be critically matched and adjusted to yield the same RF characteristics and identical audio phase and amplitude response. A combiner circuit continuously compares the RF signal strength coming into each receiver section from each antenna, and silently switches the receiver in order that the audio output is from the section receiving the strongest signal. A desirable feature is a front-panel switch for manually selecting either receiver section, which is handy when positioning antennas while checking for optimum signal strength. Automatic diversity switching should be electronic, with no noisy mechanical relays, to avoid loud clicks or pops.

Audio monitoring should be available via a headphone jack and a separate level control. The monitor audio should be independent of the main audio output, which is available through a transformer-isolated XLR connector. The XLR output should be switchable to drive professional 600-ohm line inputs (at 0 dBm nominal level) or 150-ohm microphone inputs (at −52 dBm). A floating, low-impedance XLR output enables long audio cables to be run to virtually any sound system with minimal hum, noise, and high-frequency losses.

COMPANDING INCREASES DYNAMIC RANGE

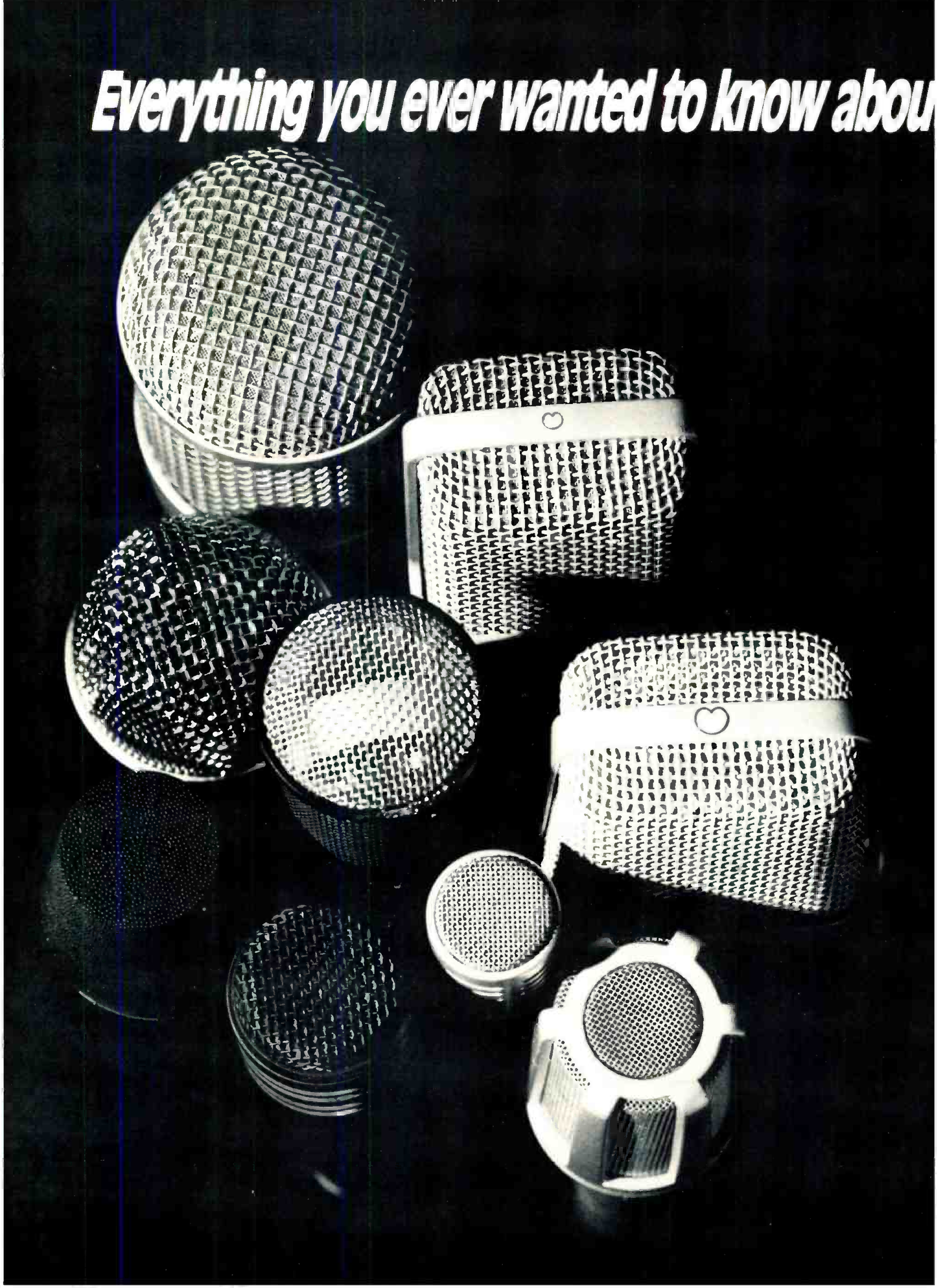
In some professional wireless microphone applications, increased dynamic range and lower residual noise are desired, and are achievable with companding, an audio signal processing technique sometimes called "dynamic expansion" or "DYNEX" (a registered trademark of Cetec Vega).

A compandor consists of a volume compressor in the transmitter and a volume expander in the receiver. (See FIGURE 2.) The transmitted volume range is compressed to approximately half the applied range (that is, by about 45 dB), by providing more gain to low-level speech phonemes than to high-level phonemes. Therefore, low-level phonemes are transmitted at a much higher level, without being masked by noise. The signal is well above the approximately −70 dB noise floor of the transmitter modulator and receiver discriminator circuits and, therefore, this noise is not detected. After the discriminator, the receiving volume expander performs the reverse function, linearly restoring the signal intensity to its original 90-dB dynamic range, while reducing received noise between syllables by approximately 20 to 22 dB.

The compander in a high-quality professional microphone system will be nearly "invisible"; that is, audio pumping will not be noticeable. However, sound mixers will be able to record very weak sounds with very loud sounds (such as gun shots) on the same soundtrack without distortion or recovery delay.

INSTALLING ANTENNAS

The installation of antennas is one of the most important factors in a wireless microphone system, for preventing fades and dropouts. In many installations, a whip antenna connected directly to the wireless receiver is adequate. However, when a large area must be covered or when signals encounter obstructions, a dipole antenna is recommended, mounted in a high location, preferably in line-of-sight with the transmitter. The antenna elements should be in a vertical configuration. For even greater distances, a 50-ohm multi-element yagi antenna,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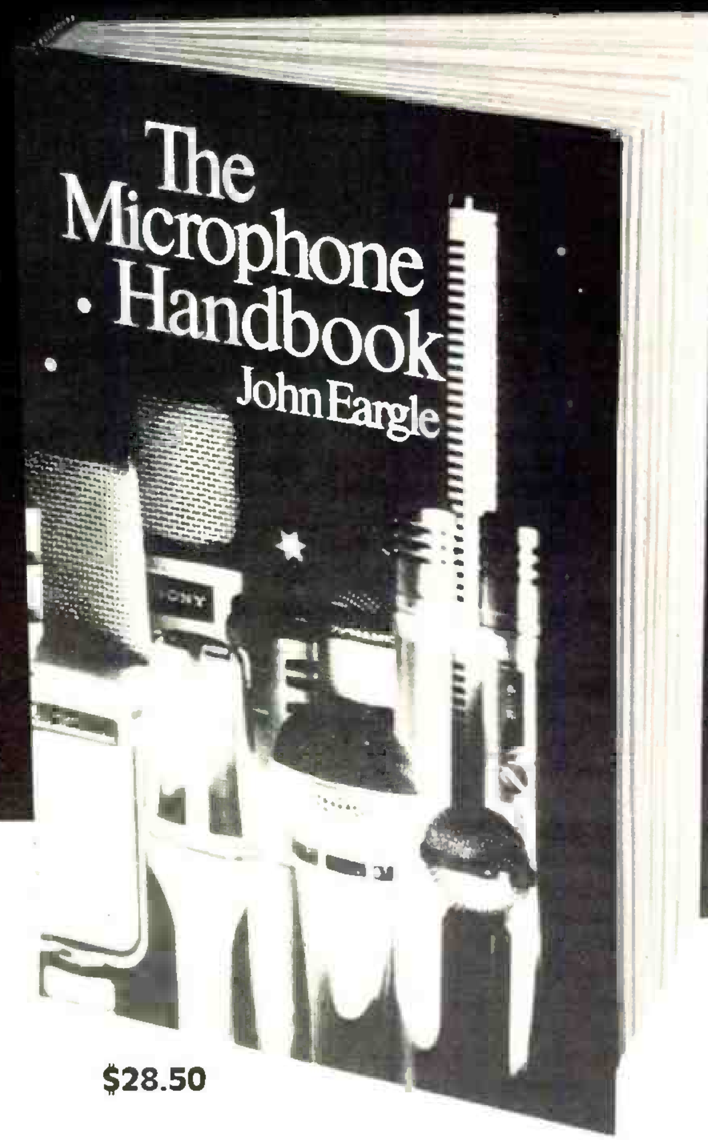

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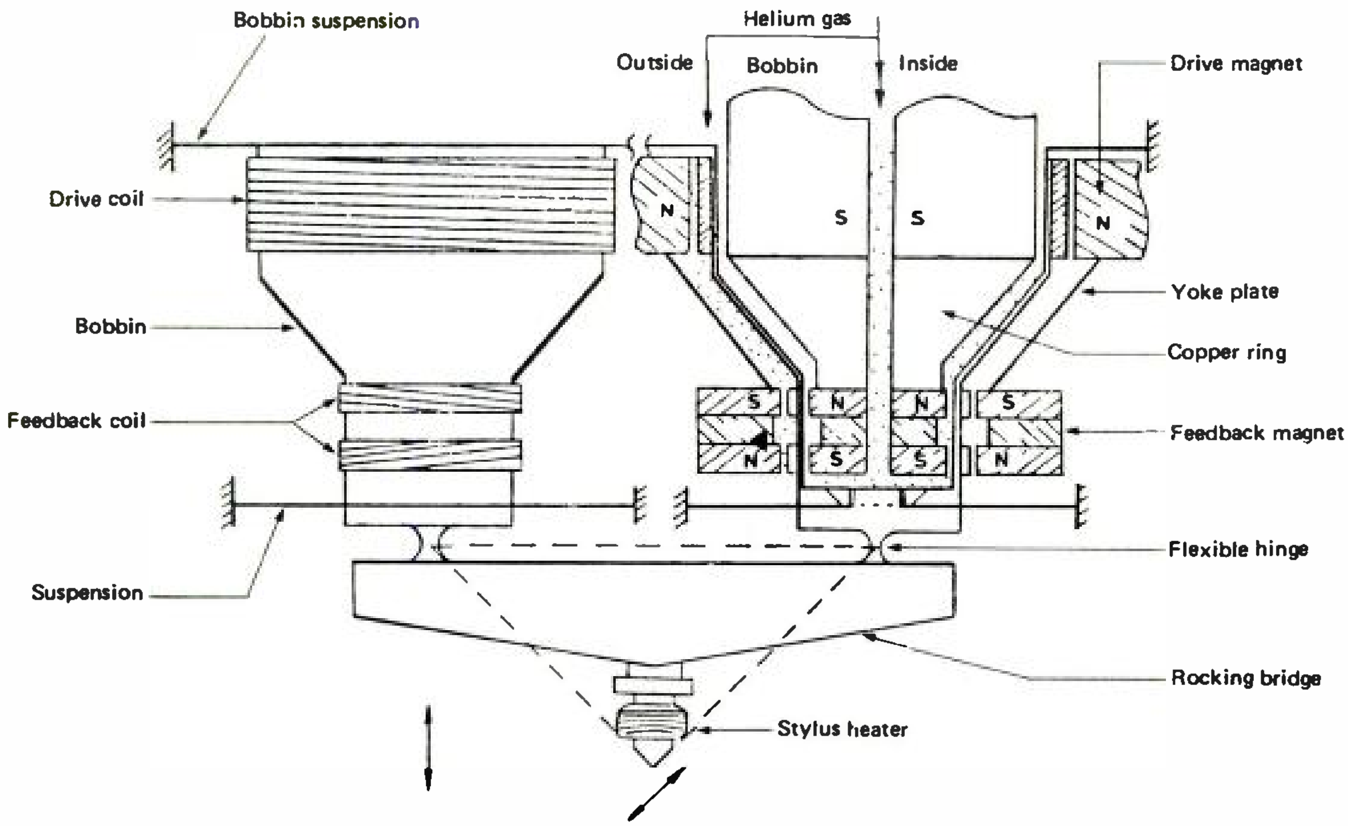
Figure 4. The JVC CH-90 Cutter Head (A), and Cutter Head construction details (B).



kHz. By contrast, 3M believes 50 kHz is still better for purely technical reasons, while conceding that "*...48 kHz is very workable, and presents no real sacrifice in audio quality.*"

Meanwhile, how would you like a digital recorder that can shuttle from any point to any other point within 50 *milli*-seconds? The EMT-450 Digiphon uses a 300 Megabyte disc stack which will store about 35 minutes of stereo program. For the moment, the system is intended for applications in which "instant access" is a primary consideration, and it will probably be some time until that storage system shrinks in size (and cost) sufficiently to compete with the traditional reel of tape. More than three years ago, producer David Rubinson asked for "...a memory core of some kind with much easier (than tape) data retrieval." (See "Automating the Automatt" in the November, 1978 **db**—Ed.) Hang on, Dave, help is on the way.

As a point of interest (and possibly, food for thought), note that the Digiphon has "fast forward" and "rewind" buttons. The system doesn't really need them, but its operator does. Although the Digiphon will find any *known* location instantly (that is, within 50 ms), some sort of tape-like forward-back-

Figure 5. Lexicon's Super Prime Time Programmable Digital Delay Processor.

ward control is still needed so that the human operator can roam through the program, searching out whatever it is that needs to be found.

On a smaller scale, JVC introduced a two-channel digital compact cassette deck. Although the cassette's size and physical format are the same as the conventional analog Philips compact cassette, the similarity ends there. Needless to say, your favorite brand of blank cassette tapes will not work in the digital domain. The system was so designed in order to take advantage of the ease of operation, portability, and already-proven production techniques of the compact cassette format, and not for any sort of A/D interchange.

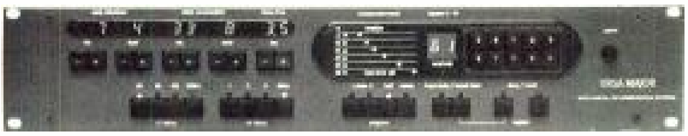


Figure 6. The Ursa Major 8 x 32 Digital Reverberation System.

Closer to the other end of the signal path, JVC introduced its new CH-90 cutter head and CA-90 cutter drive system. Some years ago, the JVC Cutting Center opened in Los Angeles. Originally intended as a half-speed cutting service for CD-4 records, the center soon attracted a following of quality-conscious engineers and producers who wanted to employ half-speed techniques for their conventional stereo productions. Over the years, JVC has customized their cutting lathes with high-torque, quartz-locked lathe motors, as well as with transformerless consoles. The cutting head and drive package are the latest refinements to the system.

THREE-CHANNEL SOUND?

At recent AES conventions– incuding this one –JVC has been showing its AHD (Audio High-Density) disc in a format comprising three audio channels and a still-frame video channel. A JVC paper cites the advantages of the third (center) audio channel for improved image stability. Although the theory is certainly indisputable, there may be some consumer resistance to an A/V system that is "neither fish nor fowl." It's likely that the consumer will find little incentive to invest in a system that offers neither full-blown video action, nor the potential for reproducing (when-and-if) quadraphonic sound.

SIGNAL PROCESSING

Naturally, digital technology continues its move into the world of signal processing. Once digital recording consoles become a practical reality, we can expect an entirely new generation of digital signal processing hardware that will not require A/D-D/A conversions at each end.

Lexicon's model 97 Super Prime Time is described as a programmable audio processor which allows the user to create, store in memory, and recall any of 32 user-created effects, plus 8 effects programs preset by Lexicon. These include two flanging programs, doubling, tripling, chorusing, and three echo programs (slap, short and long). Furthermore, the user can store his own special effects programs on a cassette tape rather than within the 97's own (non-volatile) memory. Presumably, this will protect your hit sounds from falling into enemy hands, and furthermore, you can bring them to another studio (providing they have their own 97). For those who may feel a trifle more secure, the effects programs could be recorded on the multi-track master, just ahead of the tune, and loaded into the 97 at the beginning of the mixdown. Now, all that remains is to figure out a way for bringing the effects under software control. Then the studio automation system could "patch in" the desired effect as needed during the mix.

Ursa Major's new 8 x 32 Digital Reverberation system contains 64 non-volatile registers in which the user can create a variety of reverberation effects, with decay times of use to 20 seconds. Early reflections are variable between 6 and 96 ms, in 16 steps, and with 8 steps of level control. Once a desired effect is established, it can be stored in any user-designated register. Later, the effect may be recalled, modified if desired, and re-stored in the same, or in a different, register. The 8 x 32 made its public debut earlier this year, in the sound reinforcement system used during the Simon and Garfunkel concert in Central Park on September 19th.

NOISE REDUCTION

Meanwhile, back in the analog world, Dolby Laboratories have introduced the SP series of noise reduction units. In only 12¼ inches of rack space, the SP24 will accommodate 24 channels of noise reduction, using the standard Dolby cat. 22 cards.

dbx's new model 180 provides two encode/decode channels of their Type I noise reduction in a 1¾ inch rack-mount

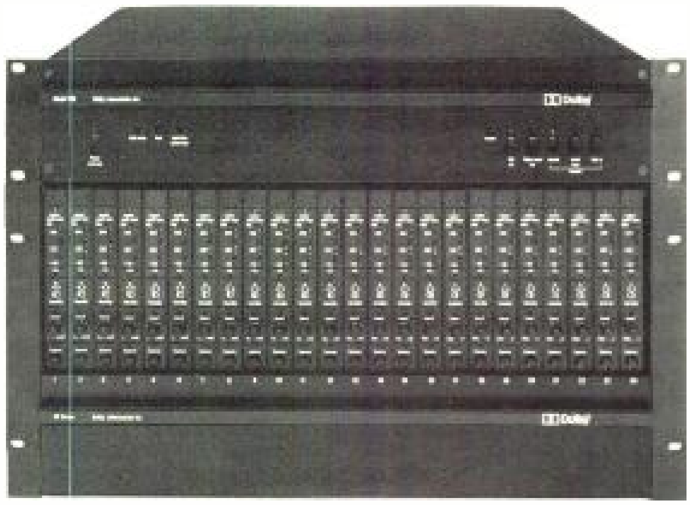


Figure 7. Dolby's new SP series accommodates 24 channels of noise reduction in one 19-inch rack panel.

configuration, and is intended for use with any two-track tape machine. The company is also entering the Dolby-dominated cassette industry. Several months ago, dbx began a licensing program, and their cassette noise reduction system has already appeared in decks from Technics, Yamaha and others. The new 2-by-2½ inch noise reduction cards are expected to attract more manufacturers, according to a dbx announcement, which notes that the same system will also reproduce dbx-encoded discs.

Press reaction to the CBS CX noise reduction system for discs has been—to put it mildly—less than enthusiastic. (No comment from us here at **db**, since we really haven't heard much of the system yet.) Part of the backlash has been directed at

CBS' claims of "compatibility" ala Dolby-B, in which CX-encoded discs are said to playback satisfactorily, even without CX decoding. Some critical listeners say this just isn't so, and that while CX discs may not be as bad in their undecoded state as dbx discs, they are certainly not "compatible" with playback

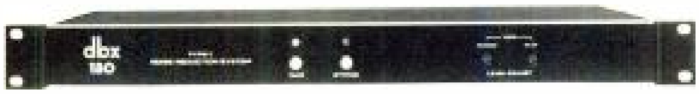
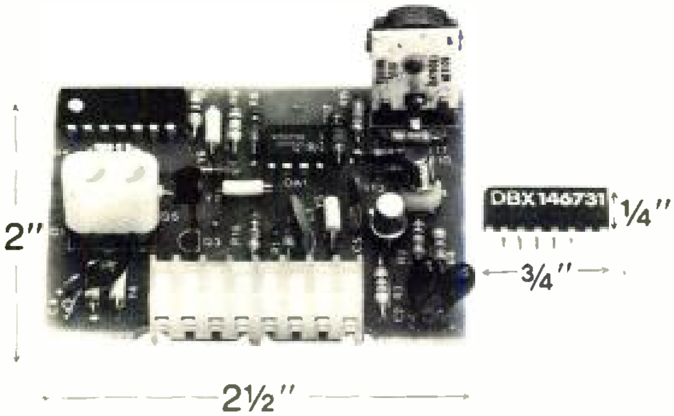
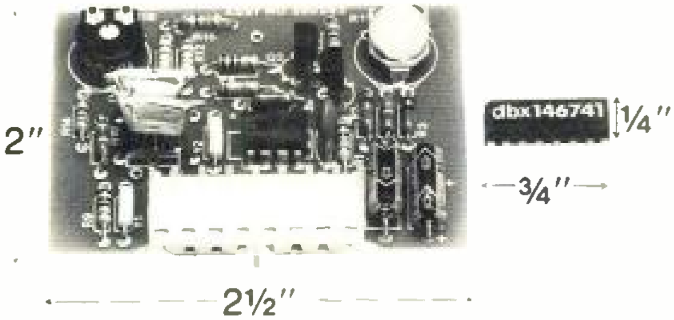


Figure 8. The dbx model 180 Type I Noise Reduction System (A) and the noise reduction cards for cassette decks.





systems that do not yet have decoding available.

For those who take their noise reduction seriously, the CBS Technology Center has licensed UREI as the exclusive manufacturer of the CX noise reduction encoder/decoder. Accordingly, UREI introduced its model 1181, a snappy-looking CX encoder/decoder for disc mastering. The system features preview, program and audition modes, metering and calibration controls which may be protected against "busy fingers" by a screw-on security cover. There's even a special equalization option for video disc applications.

Micmix Audio's new Dynafex is described as a single-ended noise reduction system, which may also be used as a noise gate. The system uses a voltage-controlled filter whose threshold may be varied over a range of 30 dB. Dynafex contains two channels of noise gate/noise reduction in 1¾ inches of rack space.

TAPE RECORDERS

From across the room, the new Ampex ATR-800 could almost be mistaken for the famous ATR-100, but on closer inspection it turns out to be a recorder designed for the broadcast industry. The ATR-800 features a built-in cue amplifier, and it even allows hands-on-the-reels type tape movement for editing and cueing. It is available in one-, two- and four-channel formats, with a basic two channel recorder priced at $5,450.

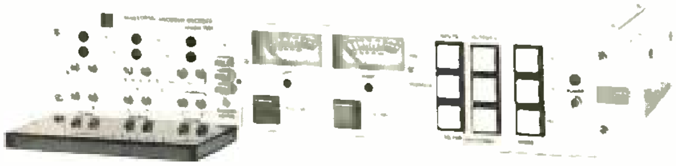


Figure 9. UREI's model 1181 Mastering Encoder/Decoder for CX noise reduction.

From Great Britain, Itam Tape Recorders are now represented here in the colonies by Itam America, Inc. The company's model 1610 offers 8 or 16 tracks on one-inch tape, at speeds of 7½, 15 or 30 ips. The 1610's roll-around cabinet contains ample space for 16 tracks of noise reduction beneath the recorder's electronics.

Otari's MTR-90 multi-track recorder has been joined by the MTR-10 series of two- and four-channel recorders. Tape transport performance is looked after by an 8080A micro-processor, and four EPROMs provide firmware control of the system's operating parameters. For easy interface to editing systems, synchronizers, etc., all transport and other functions appear at a rear-panel multi-pin connector.

TIME CODE HARDWARE

BTX's series 5000 time code products include SMPTE time code generators, readers and generator/reader combinations. Each of these is available with a video character display option,

Figure 10. The Ampex model ATR-800 audio tape recorder.

consisting of a 7-by-9 dot matrix shown against a luminance window. Size and position of the display are both user-adjustable. (BTX's Shadow System was described in detail in last month's **db**—Ed.)

Commercial Electronics, Ltd. introduced its OMNI-Q TL series of synchronizers. The TL-1 is a Sync/Effects unit which generates time code and provides a manual variable offset for phasing and flanging effects. The TL-2 Expansion Module adds LCD readout of time and a "phase" meter which displays the variable (±30 ms) offset. The TL-2 also provides tape transport controls.

The Omni-Q does not use SMPTE time code. Instead, it generates a 40-bit code (SMPTE uses 80 bits), modulated on a 21 kHz carrier. For video sync, the carrier is lowered to 12 kHz, to accommodate systems that cannot record the higher frequency carrier. In either case, the TL-2 displays minutes:

seconds: frames, but not hours. The Omni-Q literature points out that when the 21 kHz carrier is used, the time-code channel may also be used for some limited-range audio as well.

As a possible source of confusion, Conex Electro Systems also manufacture an Omni-Q. However, this one is an 8 stereo channels in/1 out general purpose cue and monitor amplifier with a suggested retail price of $450.

A newcomer to AES shows is Adams-Smith, Inc., whose series 2600 time code products are a group of modules, each dedicated to performing a specific function. Available modules include longitudinal and vertical interval generators and readers, code restorers, sync generators and display modules. By using a "building block" approach, the user may start out with a minimal system, and add modules later on, as required. Also available is the model TS-605; a Tape Synchronizer designed for one master and two slave transports.

ELECTRONIC MUSIC

The McLeyvier Electronic Music System units typewriter and piano-type keyboards, a video monitor and a graphics plotter into a complex instrument that should keep any musician/engineer busy for hours, if not indefinitely. The user

Figure 11. David McLey, his computer-based McLeyvier Electronic Music System (A), and a sample hard-copy printout from a Hewlett-Packard Graphic Plotter which was interfaced with the system (B).

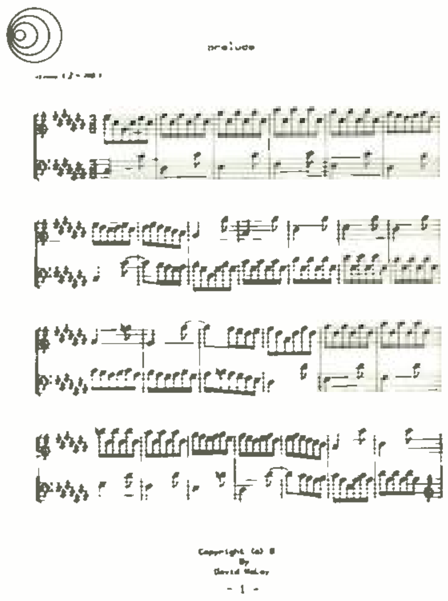


Figure 12. A few weeks after the convention, Con Brio announced its new ADS 200 Scorewriter System option, shown here with the Con Brio Digital Music Synthesizer.

plays the piano keyboard, while using the typewriter keyboard (a computer, naturally) to program the type of sound that is needed. The video monitor displays the usual instructions and prompt lines, or it may be used to display the musical notation of what has been played/programmed. Once you're satisfied with the results, the system will print out a hard-copy musical score. Individual parts may be printed, as well as a complete conductor's score. This feature alone should make it very attractive for studio arrangers, who can now wait until the last minute to finish a chart, and still get the parts "copies" in time for the session.

THE AES ANTHOLOGY SERIES

Some years ago, the Audio Engineering Society began its Anthology Series— a collection of papers previously preprinted in the Society's Journal, with each Anthology dedicated to a specific topic. The papers have been selected by an authority in the particular subject, and grouped under various sub-categories. For example, David Klepper selected 73 papers for the Sound Reinforcement Anthology, and these are presented chronologically within the following sections: General Sound System Design and Evaluation, Control of Feedback and Equalization, Delay, Loudspeaker Cluster Design, Indoor Systems, Outdoor Systems, and Circuit Considerations.

In all, six Anthologies are available at member/non-member prices of $19.00/22.00 ($17.50/20.00 for two or more Anthologies). The Anthologies may be ordered from the Audio Engineering Society, Inc., 60 East 42nd Street, New York, NY 10165. Titles, editors, number of papers, and dates are given below:

Quadraphony, J. G. Woodward, 34 papers, 1969-75.
Sound Reinforcement, David L. Klepper, 73 papers, 1953-78.
Loudspeakers, Raymond E. Cooke, 61 papers, 1953-77.
Microphones, Louis A. Abbagnaro, 63 papers, 1953-79.
Disk Recording, Volume 1: Groove Geometry and the Recording Process, Stephen F. Temmer, 73 papers, 1953-80.
Disk Recording, Volume 2: Disk Playback and Testing, Stephen F. Temmer, 83 papers, 1953-80. ■

db New Products & Services

WIRELESS INTERCOM

• Cetec Vega has expanded its wireless-intercom "Q System" to allow interfacing with wired intercom systems and to allow up to seven parties in a full-duplex wireless communications system. The new QX-1 System allows one party equipped with a QT-1/QR-1 system to be a walk-around full-duplex wireless station within an otherwise wired intercom system (such as RTS, Clear-Com, David Clark, etc.). The new QX-2 System allows from two to six parties, each equipped with a QT-1/QR-1 system, to operate within a full-duplex, party-line communications network, with an optional interface to a wired intercom system. One party can be strapped for priority if that party operates in a push-to-talk mode. Furthermore, an additional local operator may be plugged into the QX-2 base station. Accessories include antennas, headsets, leather holster, belt clip, interconnecting cables, and connectors. All transmitters are compatible with all standard electret and magnetic microphones.

Mfr: Cetec Vega
Circle 42 on Reader Service Card

STUDIO CHASSIS ACCESSORY

• Orban has announced the availability of the new Studio Chassis accessory (Model 8100A/ST) for the OPTIMOD-FM 8100A signal processing system. The new accessory chassis allows the OPTIMOD-FM system to be split into compressor and limiter sections. The accessory chassis accepts three cards removed from the main OPTIMOD-FM chassis, and is placed on the studio side of the STL (dual microwave or phone lines), preventing STL overload. Additionally, this configuration makes most operating control available at the studio. Two buffer cards are installed in the main OPTIMOD-FM chassis at the transmitter. The main chassis then performs only the peak limiter, high frequency limiting and stereo generating functions.

Mfr: Orban
Price: $795.00
Circle 43 on Reader Service Card

TEST CABLE

• A test cable for rapid connection with a double banana plug and push-on BNC male, has been introduced by ITT Pomona. Model 4767-C, available in four lengths from 609.6 mm (24 in.) to 1524.0 mm (60 in.), features beryllium copper outer fingers on the BNC, maintaining shield continuity. The double banana plug is insulated with polyethylene and can withstand temperatures up to +55 C (+131 F).

Mfr: ITT Pomona Electronics
Price: 4767—C—36 (36 in.) $9.75
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COMPUTER SOUND PROCESSOR

• The DMX-1010 is a programmable digital audio signal processor designed for music synthesis and musical signal processing. As a signal processor, the DMX-1010 will do delay, reverberation, phasing, flanging, equalization, compression, limiting and more. Synthesis may be combined with signal processing. All synthesis and processing is done digitally, with 16-bit accuracy, giving very high fidelity. All synthesis and signal processing units are implemented in software that runs on the DMX-1010's two computers—a PDP-11/03 microcomputer and a DMX-1000 signal processor. The 11/03 is responsible for the operator interface and for doing such processing as can be done at rates up to 100 Hz. The DMX-1000 is an ultra-high-speed microprogrammable audio signal processor that does processing at the audio rate. The DMX-1010 includes a CRT terminal and a real-time control panel containing 10 knobs and five switches. Specifications include audio sampling rate adjustable from 19.3 kHz to 100 kHz, with bandwidth from 0 up to 40 kHz. A/D and D/A converters have 16-bit resolution, giving a 90 dB dynamic range. Up to 21 oscillators may be used at a 19.3 kHz sampling rate.

Mfr: Digital Music Systems, Inc.
Price: $35,300
Circle 45 on Reader Service Card

STEREO MIXERS

• Available with 8, 12, and 16 input channels, the 22 Series offers some professional features not commonly found on stereo mixers. Some of the features found on 22 Series consoles include: (Input Channels) Balanced Mic and Line level inputs, Input preamp in/out jacks, Mic/Line switching, Peak LED indicators, Monitor, Reverb (all 22 Series consoles have built-in reverb featuring the Accutronics type 9 tank), Aux (with pre or post EQ/fader switching), 3-way EQ, Pan, Solo, and slide faders. Master Control features include extensive headphone monitoring and Solo system, Switchable metering, Panable Aux return and Panable reverb, Slide fader master output controls for Left/Right Monitor master and Mono. Rear panel connections include both balanced and unbalanced line level outputs on all main output functions (L/R, Monitor and Mono), High and Low level Aux return jacks, and Master function inputs that allow two 22 Series consoles to be interconnected for more input channels. In addition, the 22 Series can use its own internal power supply, or may be powered with NEI's XMP remote supply simply by throwing a switch on the rear panel.

Mfr: Neptune Electronics, Inc.
Circle 47 on Reader Service Card

Circle 46 on Reader Service Card

DISSOLVE STAND AND TRAVEL CASE

• A new 2-level stand which will hold two Kodak Ektagraphic or Carousel slide projectors has recently been introduced by Tiffen. The new stand provides control knobs for adjustments of the horizontal and vertical positions of the projectors. The model D2 Stand is available either as a separate unit or as a part of the Model D1000 Traveling Showcase, which stores Stand, holding projectors and trays. The D1000 also features space for additional lenses and storage for dissolve units.

Mfr: Tiffen Manufacturing Corporation
Price: D1000—$335.00, D2—$139.50
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Circle 38 on Reader Service Card

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People/Places/Happenings

• **Vladimir G. Nikanorov** was named vice president of engineering for **Muzak**, according to **Edward J. Fuhrman**, president. Mr. Nikanorov assumes full responsibility for all of the current engineering operations, as well as the implementation of anticipated technical advances. Mr. Nikanorov comes to Muzak from **Bonneville Broadcast Consultants** where he was technical director, providing engineering supervision for the 100-plus radio station clients of that operation. His new appointment is the culmination of more than 17 years of experience in the field of broadcast audio engineering both in this country and abroad. Prior to joining Bonneville, Mr. Nikanorov was with **George Thomas & Associates**, Los Angeles, as a consultant in audio visual systems. He has also been associated with **Schulke Radio Production Studios**, the country's largest broadcast syndicator of "beautiful music," and the **Vanguard Recording Society** in New York City.

• **Fostex Electro Acoustic Systems**, a division of **Interlake Audio Inc.**, announces the appointment of **Paul M. Gardocki** as vice-president, marketing and development. Gardocki comes to Fostex Electro Acoustic Systems from **W.E.D. Enterprises**, a subsidiary of **Walt Disney Productions**, where he was involved in sound systems design and loudspeaker evaluation for Disney's E.P.C.O.T. project. Gardocki's previous experience includes marketing positions with **Emilar, Marantz and Cerwin Vega.** Gardocki's responsibilities with Fostex include supervision and management of the company's marketing programs and sales force for the Fostex Laboratory Series monitor systems, printed-ribbon microphones, studio headphones and audio power amplifiers.

• **Dave Harrison**, president of **Harrison Systems, Inc.**, the Nashville-based manufacturer of audio mixing consoles, has announced a reorganization of its domestic marketing which will affect all product sales and support within the United States. Harrison is changing to a direct-to-end-user sales policy. In most parts of the United States, Harrison customers will continue to do business with the same people as before. The difference is that now these people are factory representatives rather than dealers. They now receive compensation from Harrison Systems rather than buying equipment and reselling it at a profit.

• **S. N. Shure**, Chairman of the Board of **Shure Brothers Inc.**, Evanston, Illinois, has announced the election of **J. H. Kogen** to president and general manager. Mr. Kogen's previous position was executive vice president of operations.

• **3M** has added **Roger Harvey**, sales representative, to its digital audio equipment team for a greater concentration of sales activities in the Southeast. Based in Atlanta, Harvey will cover Tennessee, Mississippi, Alabama, Georgia, Florida, and South Carolina. This territory was formerly covered by **William McNutt**, who is now concentrating his digital sales activities in the Southwest, based in Dallas. Harvey's background includes professional audio recording experience at the studio level, as well as radio, television and video production work. He also has sales responsibility for 3M's video recording and related video products in southern U.S. The 3M Digital Mastering System is the first commercially available 32-track recorder. 3M also makes a four-track recorder and an electronic digital editor.

• **Trident** (U.S.A.) announce the sale of their Series 80 consoles to various studios in the U.S.A. **A & R Recording** in New York have purchased a Series 80 with **Melkuist** Sub-Group Facilities. Melkuist also boasts a V.C.A. By-Pass switch on the fader enabling it to be utilized as a regular log fader. A & R are the second room in New York to order the Trident/Melkuist combination, the first being **Record Plant. Mayfair Sound** of Manhattan has also purchased a Series 80 and T.S.R. 24 Track Tape Machine for installation in their Studio "A," which is undergoing considerable renovations.

• **Quad-Eight Electronics** has announced that the second of two disk-automated **Coronado** editing systems has been delivered to the **National Broadcasting Company's** Burbank facilities. The Coronado systems will be installed in NBC's new video "sweetening" studios, scheduled to go into operation early in 1982. Each system uses Quad-Eight's automated, 40 input Coronado console and proprietary "Compumix III" software package and includes dual disk drives, color graphics terminal and Compumix editor computer. Using SMPTE time-based synchronization, the Quad-Eight system provides a tool for audio enhancement, with mix editing accuracy down to a single video frame. A companion system, the Ventura II editing system, was featured by Quad-Eight at the recent Audio Engineering Society show in New York.

• **Bose Corporation** has announced the appointment of **Richard Tyler** to the position of sales assistant for its Professional Products line in the eleven Western states. Tyler has been involved in the sound industry since 1976. He has served in a retail sales capacity at **Pacific Stereo** and in field sales positions for **Boman Industries and Cytec Electronics.**

• **Terry Sorensen** has been appointed product applications engineer for **James B. Lansing Sound, Inc.**'s International Division, it was announced by **Bruce Scrogin**, division vice president. As product applications engineer, Sorensen provides a variety of technical and sales support services to the International Division, such as consulting on professional installations and directing dealer orientation sessions for new equipment.

• **Allen & Heath Brenell, Ltd.** of London, England, announces the formation of **Allen & Heath Brenell USA, Ltd.** to handle all operations of the firm in the United States. Appointed to the position of national sales manager is **Mr. Charles Augustowski**, a long-time representative of Allen & Heath products in the U.S. Among his first activities includes reorganization of AHB operations in the U.S., tighter quality control of all products shipped from the Stamford CT headquarters, and increased dealer and rep support. An expanded dealer network will be established with the appointment of new regional representatives.

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