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# SOUND & COMMUNICATIONS

FOR CONTRACTORS, SYSTEM MANAGERS AND SPECIFIERS MARCH 1987

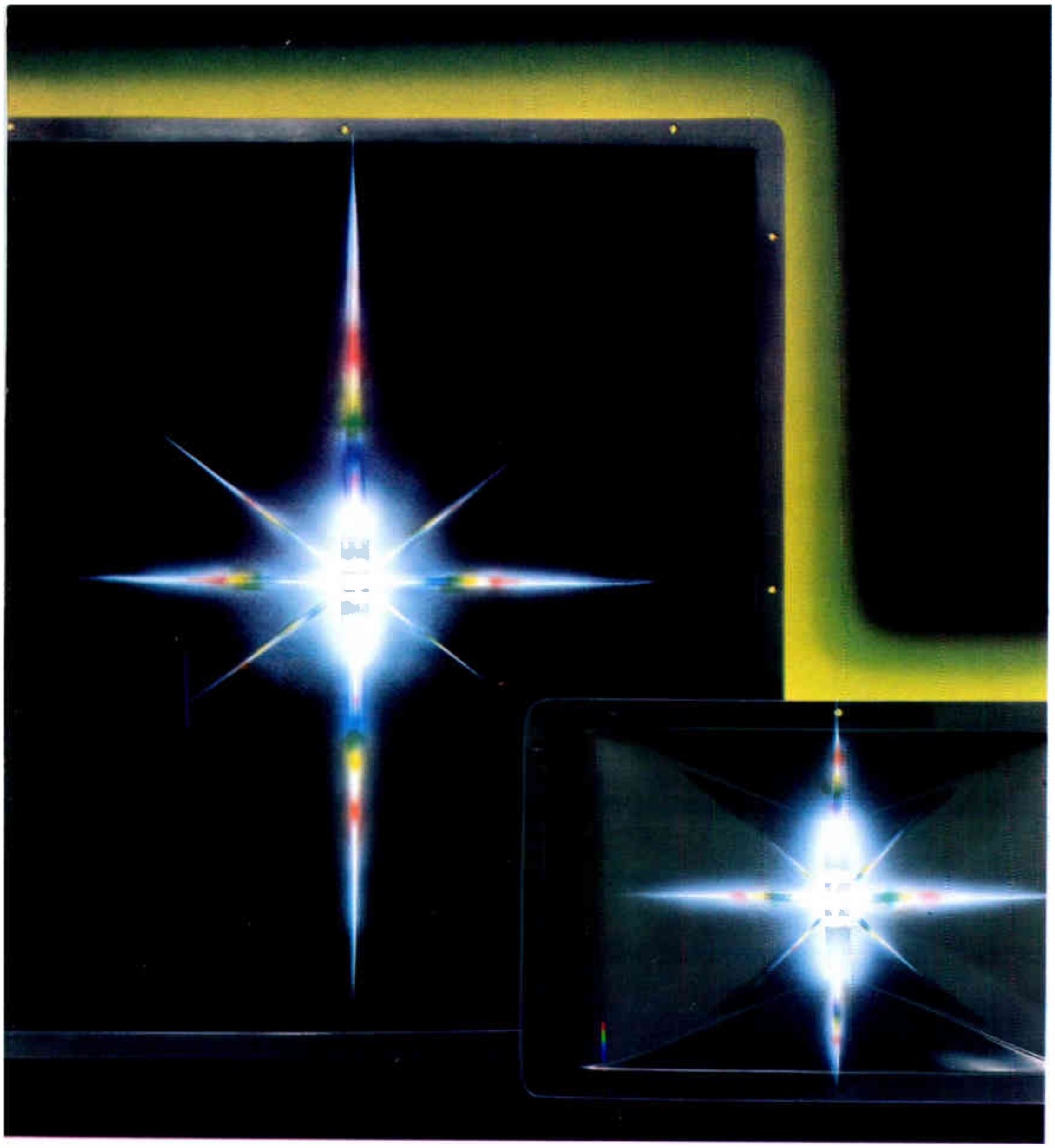
## The DSP Chip



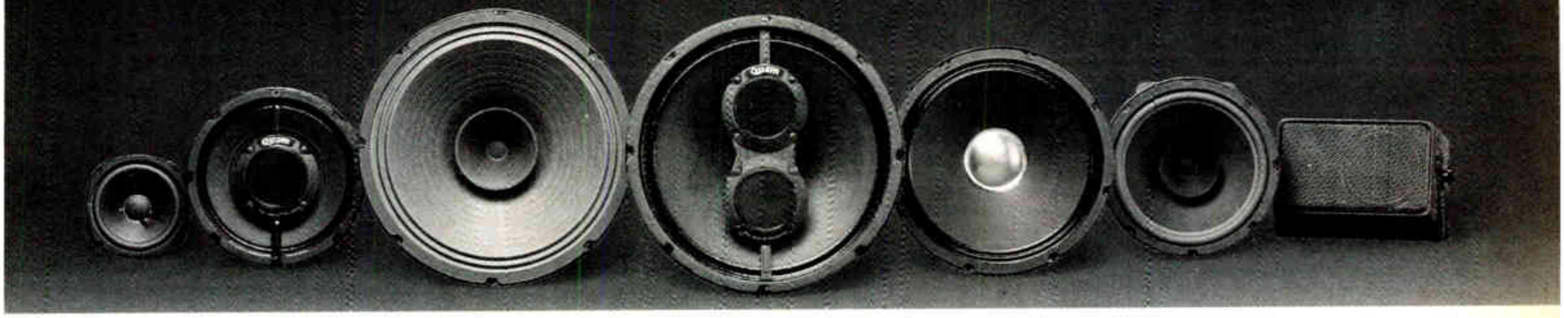
*Profile:* The Combining System at the Orlando Marriott

Diversity Antenna Systems

*Lab Test:* BES C-70S Loudspeaker



# YOU KNOW US FOR SPEAKERS.



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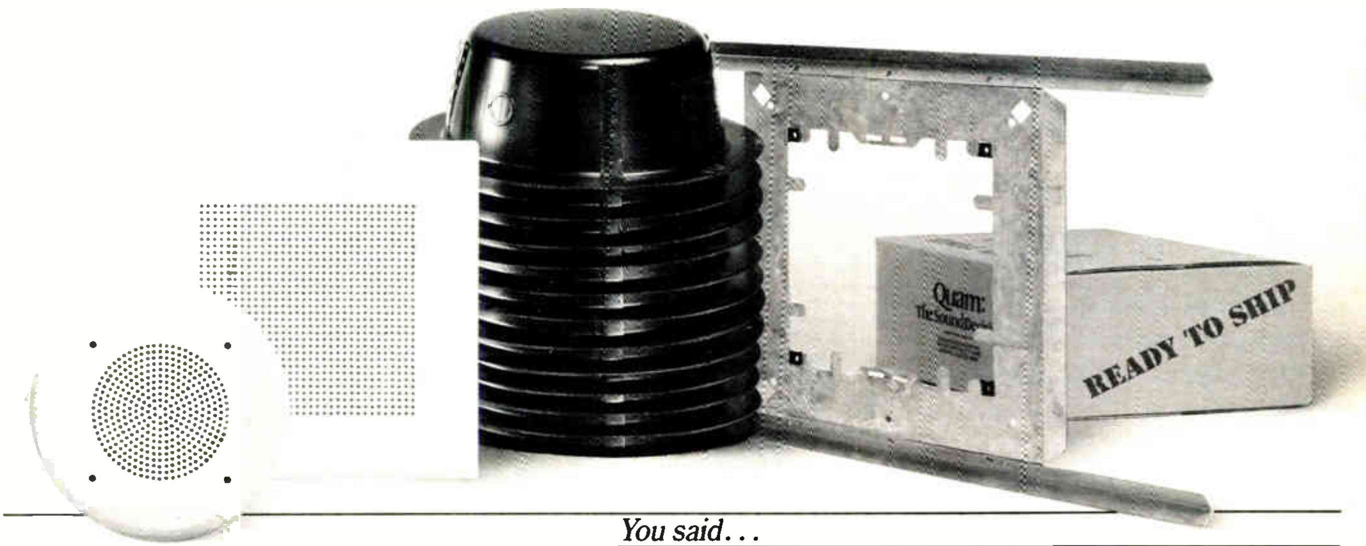


*You said. . .*

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*So while speakers started it all, listening keeps us–and you–prospering. And we'll continue to listen–and respond to what we hear–while we refine and enhance our current products and create new ones to meet your evolving needs.*

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# SOUND & COMMUNICATIONS

Volume 33 #3
March 1987

24

## FEATURES



## COLUMNS



## DEPARTMENTS



**ON THE COVER**
The TMS32010 DSP chip from Texas Instruments.

combining **SOUND MERCHANDISING & MODERN COMMUNICATIONS** FOUNDED 1955 
Sound & Communications (U.S. P.S. 943-140) is published monthly by Sound & Communications Publications, Inc., 220 Westbury Ave., Carle Place, N.Y. 11514. 516-334-7880. President, Vincent P. Testa. Subscription rates: U.S.—3 years $37.50, 1 year $15.00. Canada and Mexico—3 years $48.00, 1 year $18.00. Foreign and South America—1 year $25.00. Second-class postage paid at Carle Place, NY and at additional mailing office. POSTMASTER: Send address changes to Sound & Communications, P.O. Box 159, Carle Place, NY 11514.

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Circulation audited by
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The Music & Sound Buyer's Guide • Post
NSCA-TV news • NAMM-TV News • CES-TV News
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# "When Cunard wanted sound fit for a queen, I specified Bose."

– Peter H. Bird of Bird Brothers Communications Ltd., Oldham, England, contracted by Cunard Line to design and install sound systems for their flagship Queen Elizabeth 2

It's hard to imagine a more challenging place to install a sound system than the Queen Elizabeth 2. After all, she *is* the world's most renowned luxury liner, and people enjoying her hospitality naturally have great expectations. They expect to dine, play and relax on their cruise in style and comfort. *And almost everything on board the Queen, including the sound system, is important in cultivating the right atmosphere.*

But the QE 2's sound system has to do more than sound good. Should the need arise, it also has to be capable of providing passengers with vital safety information. It must meet the strict standards set by a number of organizations including Lloyds' Registry of London, the U.K. Dept. of Transport and the U.S. Coast Guard. It needs to be rugged enough to withstand environmental extremes, easy to install, and above all, it has to be reliable—because the nearest service may be an ocean away.

Bose professional sound products.

So when Cunard Line was looking for such a sound system for the QE 2, they turned to Bose for quality and durability they could rely on. Bose professional products offer exactly the right combination of high fidelity, ruggedness and port-to-port reliability that a floating resort city needs.

Even if your particular sound application isn't a floating palace, guests still deserve to be treated like royalty. Discover the sound systems truly worthy of a queen by evaluating Bose professional products soon. For more information, write Bose Corporation, Dept. SC, The Mountain, Framingham, MA 01701.


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**CETEC CORP. CONSUMMATES ACQUISITION OF ORIGINAL CYLINDER CO., INC.**

Robert A. Nelson, president and C.E.O. of Cetec Corporation has announced that Cetec had consummated the acquisition to purchase the assets of Original Cylinder Co., Inc. in an all cash transaction. The purchase price was not disclosed. Original Cylinder Co., Inc., located in Los Angeles, CA, is involved in the rebuilding of replacement parts for the automotive after market. For the fiscal year ending October 31, 1986, Original Cylinder had sales of approximately one million dollars.

Nelson indicated that the acquisition of Original Cylinder Co. reflects Cetec's strategy of purchasing companies in niche markets which are capable of earning a higher rate of return on invested capital. Nelson also announced that the previously announced preliminary agreement by which Cetec would purchase the outstanding stock of Mekel Engineering Inc. has been terminated by mutual agreement of the two companies.

**AUDIO-TECHNICA PRESENTS PRO SOUND SALES AWARDS**

Awards for sales achievements were presented by Audio-Technica president Jon R. Kelly and national sales manager Mark Taylor at the company's annual pre-NAMM sales meeting in Anaheim. Cited as outstanding for their contributions to the sales results were six Audio-Technica reps.

The Representative of the Year Award, the Order of the Samurai, was presented to Joe Blacker, president of Audio Associates, Glenwood, MD for the second consecutive year. The award is presented to the sales rep firm providing best overall support of the line, improved market share, customer relations, and professional handling of daily selling situations. Audio Associates represents Audio-Technica in Pennsylvania, West Virginia, southern New Jersey, and Delaware. Also receiving awards for over-quota sales performance were Alan Brothers of David H. Brothers Company, Joe Blacker of Audio Associates, Dick Geisler of Cambridge Marketing; Marcus Johnson of Marcus Johnson Associates, Casey McWilliams of CM Sales, Randy Fuchs of Crescendo Associates and Bill Ray of William J. Ray and Associates.

**JBL SPEAKERS USED TO SIMULATE ENGINE NOISE**

A total of 50 JBL Professional 2225 15-inch low-frequency transducers loudspeakers were used in a test at the McDonnell Douglas anechoic chamber. The testing was in part of a three-year program at the Douglas Aircraft company which is preparing to market Ultra High Bypass engines which utilize multi-bladed, counter-rotating propulsors to reduce fuel consumption. The JBL speakers were used to simulate engine noise levels to help determine methods for meeting or exceeding current airline cabin noise standards.

**ISCET PUTS A STOP TO CERTIFICATION SCHEME**

Stan Haring of Olympia, WA has agreed to a permanent injunction by the State of Washington which prohibits him from offering a certification program under the auspices of an organization he called the American Association of Certified Electronics Technicians (AACET). In a consent decree signed by both parties, Haring agreed to pay a civil penalty to the State of Washington and to desist from any future violation of state law. Haring has been an instructor at Puget Sound Community College in Olympia, WA.

The office of the International Society of Certified Electronics Technicians (ISCET) became aware of AACET when industry people mistook AACET's policies for those of ISCET. ISCET members and certification administrators advised ISCET of the AACET plan to offer certification for a fee without qualifying examinations to instructors, military and other technicians working in the industry. The AACET brochure offered instructors who purchased the $25 certificates the opportunity to "...make extra money ($10 for each applicant you test) as an official AACET examiner."

ISCET made the initial inquiry to the Washington Attorney General and cooperated fully in the ongoing investigation. Jim Parks, CET, ISCET chairman said that, "ISCET recognizes the right of other organizations to offer certification programs, however, they have an obligation to insure that their testing procedures are meaningful and that the designation instills public assurance of minimum level of technical competence." The test offered by Haring was reviewed by ISCET and Don Hatton, director of product services for the Consumer Electronic Group of the Electronics Industries Association (EIA/CEG). In the opinion of both, it was in no way an adequate measurement of the complexities of professional servicing.

## PIRELLI CABLE ACQUIRES JACOBSON BROTHERS INC.

Pirelli Cable Corporation has announced acquisition of Jacobson Brothers Inc. a leader in underwater cable laying and marine support work. The Seattle-based company owns and operates equipment which makes it possible to lay power and communications cable in all types of marine environments, including areas inaccessible or expensive to reach by conventional cable-laying ships. Pirelli, which is engaged in the design, engineering and installation of power and communications systems, plans to operate Jacobson as "a separate and distinct business headed by its existing management," according to O.F. Raimondo, Pirelli Cable president and chief executive. "Jacobson will continue to serve oil companies, power and communication utilities and the Federal and state government," he said.

## COLONIAL WIRE AND CABLE JOINS ANIXTER

Anixter Brothers Inc. has acquired Colonial Wire & Cable it was announced by John Pigott, president of Anixter. Colonial is a distributor of electronic and mil-spec wire to the OEM, Computer, and Aerospace/Defense markets on a national basis. The company will now operate under the name of Anixter-Colonial and will become part of Anixter's network of on-line, computer linked distribution centers. Jim Dubovick and Dave Ahearn started the business in 1978. They will continue with Anixter in the management of Colonial, which is equipped to dye, stripe, twist, etch wire, and perform hot stand tubing.

## SHURE ANNOUNCES FORMATION OF HOME THEATER SOUND DIVISION

Shure Brothers, Inc. has announced the formation of a new division, Shure Home Theater Sound. The new division will be responsible for the development and marketing of a line of home audio/video components to supplement Shure's HTS 5000 Surround Sound Decoder. According to James H. Kogen, president of Shure, the company has developed a long term plan for Home Theater Sound Products and has committed itself to maintaining it's leadership role in this area.

"As a long-time leader in this design and manufacturer of high fidelity and sound reproduction equipment, Shure is well suited to take a 'ground-breaking' role in bringing the theater sound experience into the home. Our initial step into Home Theater Sound, the THS 5000, has been well received from the standpoint of both consumer and professional acceptance. The primary purpose for making this organization change is to provide a clear focus for the Home Theater Sound product line and to enable those involved in that division to put 100 percent of their time into making their efforts successful." Robert B. Schulein, formerly Shure's chief development engineer, has been promoted to general manager of the new division. He has been with Shure since 1966.

by R.W. Sharer

# Sales Force Compensation

It's important to our discussion that we define our terms. When we speak of a sales force, we mean one or more *employees* of your company with direct customer account responsibility. These people may travel to customer sites to sell and promote your products and services. They may be "inside reps" whose principal job is telephone soliciting. They may even be retail store sales reps who handle customers that come to you. The bulk of the discussion is aimed at the territory sales rep in the field, but the concepts may apply to others as well.

Compensation is defined as the total remuneration paid to a sales rep either directly (as in commission, salary or bonus) or indirectly (as in use of a company car, local club membership dues, or dependent dental insurance). If you or your company incurs a cost to provide it, it should be included in your definition of compensation—whether it goes directly to the rep in cash or simply makes him or her the beneficiary of some service with value (such as paying a rep's tuition at a NAVA/ICIA Institute).

Compensation generally comes in one of five forms. First, there is *direct pay*. You can call that pay by any of several names, each of which we'll treat separately, but it is the gross dollars that an employee receives for contributing his or her services for a day, a month or a year. Internal Revenue Service calls it "wages, salaries, tips, fees, commissions, etc."

Second, there are *reimbursed expenses, expense allowances*, or *company-paid expenses* on behalf of the employee. They are a very important part of total compensation because your sales rep must incur expense to generate income, and it has become common business practice for the *company* to absorb most of them. When these equal the actual amounts expended, they are generally nontaxable to the rep, and deductible for you as legitimate business costs.

Third, there are *value-added benefits*. Condition yourself to call them value-added rather than "fringe" benefits, because they are anything *but* a fringe. And since most of them receive favorable tax treatment for you *and* the rep, they are indeed, value-added. We'll deal with these in more detail in a later section.

Fourth, there are perks. The term comes from the word "perquisite," meaning something incidental made from employment above salary. Though perks are often associated with executive compensation and include things like a leased wardrobe (yes, there is such a thing) corporate jets, company-provided tax preparation, etc., they would include—for the regular working troops—a company party or dinner, coffee pot, employee lounge or sponsored softball team.

Finally, there are *non-financial forms* of compensation. These would be left out of our discussion as having little dollar value were it not for their importance to the psychological well being of an employee. There is little financial value to an award, a trophy, or a plaque—let alone an article in your newsletter or the local paper, but its recognition value may be very significant. Allowing a rep to select his or her own company car within a specified range—or even a letter of commendation—goes far beyond its cost in compensation terms. Assuming the traditional forms of monetary programs are appropriate, these non-financial incentives often spur the rep to that extra effort that sets winners apart from others. Many companies, unfortunately, pass up these inexpensive—or free—"motivators" for reasons not easily understood.

**What's different about sales compensation?** So far, little has been said about how and why sales compensation is more complex than that of other employees. All employees expect (and deserve) fair pay, expense reimbursement, value-added benefits, "perks," and non-financial rewards. So why is it different for sales reps?

There are several, very important, differences. They trace to the environment of a sales rep, (the nature of the job) the nature of the person, and the dynamics of your company goals. Let's look at each.

**The sales environment.** Selling is lonely. Most sales reps—especially outside people—spend a lot of time alone, without the camaraderie of teammates, colleagues, or managers. There are principles of selling that increase the odds for success, but there is no "paint-by-numbers" formula to follow for getting results. A tremendous amount of creativity and personal motivation is required. Twenty times a day, the sales rep makes a choice about what to say or do next. Right choices produce good results, wrong ones bring frustration, and usually, there's only one chance at it. Often, the choices are made in an instant, with little time to ponder alternatives. On any given sale, it's usually all or nothing: you get the order or you don't. And there's nowhere to hide. No substitute for results.

Sales reps take abuse. It's seldom violent, but they do get put off, put down, or put *out* by customers. It takes extra effort then to knock on the next door.

In short, nearly everything that happens—or doesn't—comes back to the rep as either credit or blame. There can't be many letdowns or bad days.

**The sales personality.** It's neither easy nor wise to try to generalize too much about the "sales personality," because successful sales reps come in every size, shape, sex, color, and temperament. The successful ones, though, do have certain characteristics in common. (Actually, studies have shown that reps

*(continued on page 12)*

# The Latest Page in Audio History.

## 1877: The microphone is invented.

Developed by Alexander Graham Bell, Thomas Edison and Emile Berliner, it was patterned after the human ear itself. The first of many attempts to capture sound as we really hear it—a goal that took more than a century to realize.

## 1896: The first synthesizer.

Thaddeus Cahill's Telharmonium weighed 200 tons! A touch-sensitive keyboard drove a complex labyrinth of motors, pulleys and alternators.

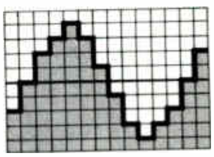
## 1924: The dynamic loudspeaker.

The design first developed by Chester W. Rice and Edward W. Kellog has changed very little over the years. But today's broad frequency bands and increasingly complex audio signals are challenging the loudspeaker like it's never been challenged before.

## 1925: The vacuum tube amplifier.

The collective work of Edison, John Flemming and Lee DeForest. Transistors later came to replace tubes, but audiophiles have never been entirely satisfied with what they heard.

## 1958: The advent of digital.



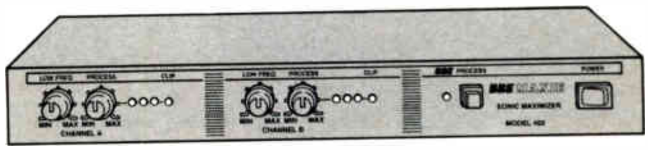
Working at Bell Telephone Laboratories, Max Matthews developed a computer program for creating and storing audio waveforms as digital data. Today, digital technology is widely available to musicians and consumers through innovations like user sampling devices and CD players. To hear the sound, however, it's still necessary to translate it back into the analog domain. And that's where problems develop.

## 1978: The BBE breakthrough.

When you put a power amp and a loudspeaker together, something has always been lost in the interface. That's where phase and amplitude distortion develop, due to "miscommunication" between amp and speaker. And that's why amplified sound has never had the dimension, depth and realism that the human ear can hear all around it in nature. That is until Bob Crooks made an important discovery—BBE. BBE is the vital "missing link" between amplifier and speaker. It analyzes the action of both—automatically and on a continual basis. It applies the phase and amplitude correction that's needed to make the sound come through the way you and nature intended it. The difference is easy to hear. Improved low-end definition and punch. Cleaner high-end transients. Better mid-range presence. *In short, unprecedented clarity.*

## 1984: BBE on stage.

Major P.A. companies like Stanal Sound and Best Audio made BBE part of their touring systems. And when the entertainment industry



gathers for such events as the Grammies and the Academy Awards, BBE is there, making sure the sound is as special as the occasion itself.

## 1985: BBE in the studio.

Award winning producer Steve Levine joined forces with the Beach Boys and teamed them up with BBE for an all-digital recording session for CBS/Caribou. *"BBE is to digital what equalizers were to analog,"* said Levine. *"I can't imagine ever recording without BBE again."*

## 1986: BBE today.

Wherever sound is amplified, recorded or broadcast, there's room for the BBE improvement. Because at the end of the line, we still have the loudspeaker that made big news back in 1924. Which is why you need BBE more than ever. Visit your BBE dealer for a demonstration. To find a BBE dealer near you call today toll-free at 1-800-233-8346. In California, 1-800-558-3963. And start making some history of your own.

**BBE** All the sound you've never heard.™
**Barcus-Berry Electronics, Inc.**

## BUSINESS FRONT

*(continued from page 10)*

who *fail* at the job have more in common than do reps who *succeed*. That is, in any group of sales reps it is easier to explain why they did *not* perform than it is to determine how they *did*.)

By and large, good sales reps are self-starters. They quickly learn that they control most of their own destiny, that they must "get on with it," that they are only as good as their last performance. They are also, generally, pretty hard workers. In their unstructured, unsupervised environment, they know that time and effort invested will be significant to achieve the results they want. The best sales reps are also good managers: well-organized, and careful about their use of resources.

Contrary to popular opinion, most are quite conservative. Sales reps are often seen by others as easygoing, carefree, and free-wheeling. In reality, most are quite security-minded, personally intense, and much more prone to caution than most observers believe. Though a few opt for a flamboyant lifestyle—and convey a devil-may-care attitude, this is often a mask for certain insecurities and/or a defense against the constant threat of rejection and failure in a very high-exposure job. It's also an acquired skill that makes people like them (an important asset) and a demonstration of their adaptability to many and varied—sometimes high-pressure—situations.

People are important to them. Not just for obvious business reasons, but because they genuinely enjoy working with and being around people. They can get along with almost anybody—if they want to.

Taken together, then, we have a cautious (perhaps insecure), hard-working, people-oriented, courageous individual who lives in a world that doesn't allow him or her to fulfill any of their most driving needs very easily. They have something to prove every day. To you, to the customer, to the world, and mostly to themselves.

Little wonder, then, that appearances are important to them. That they sometimes act like Prima Donnas and want to be treated like it. That they *insist* on recognition and reward for achievement. That compensation is so important a yardstick, and that they are so difficult to manage.

**Your changing goals.** No business environment remains static for very long. We see the ebbs and flows of the A-V market as a clear indication that change is accelerating. Our "bread and butter" isn't what it used to be, and in fact, many of us have to scramble just to stay afloat. Nowhere is this more sharply-focused than with your sales force. They are the instruments of change—and sometimes the victims of it. When you take on a new line, rearrange territories, start a new program, or revise a pricing structure, you expect the sales force to respond by changing their emphasis, supporting your efforts, achieving *different* goals.

Because they are goal-oriented as a rule, they can do it. Because they are loyal and hardworking, they usually *will* do it. But management must recognize that it causes a number of problems for a sales rep to shift marketing gears, and you must see why. Today's sales harvest is the result of last year's planting. To plant a new crop is to have to wait again for the harvest—and the money you'll get for selling it. New planting diverts attention from

*(continued on page 60)*

**R-42 diversity receiver • Now with GaAsFET's.**

# Improved sensitivity and system range, with ultralow noise.

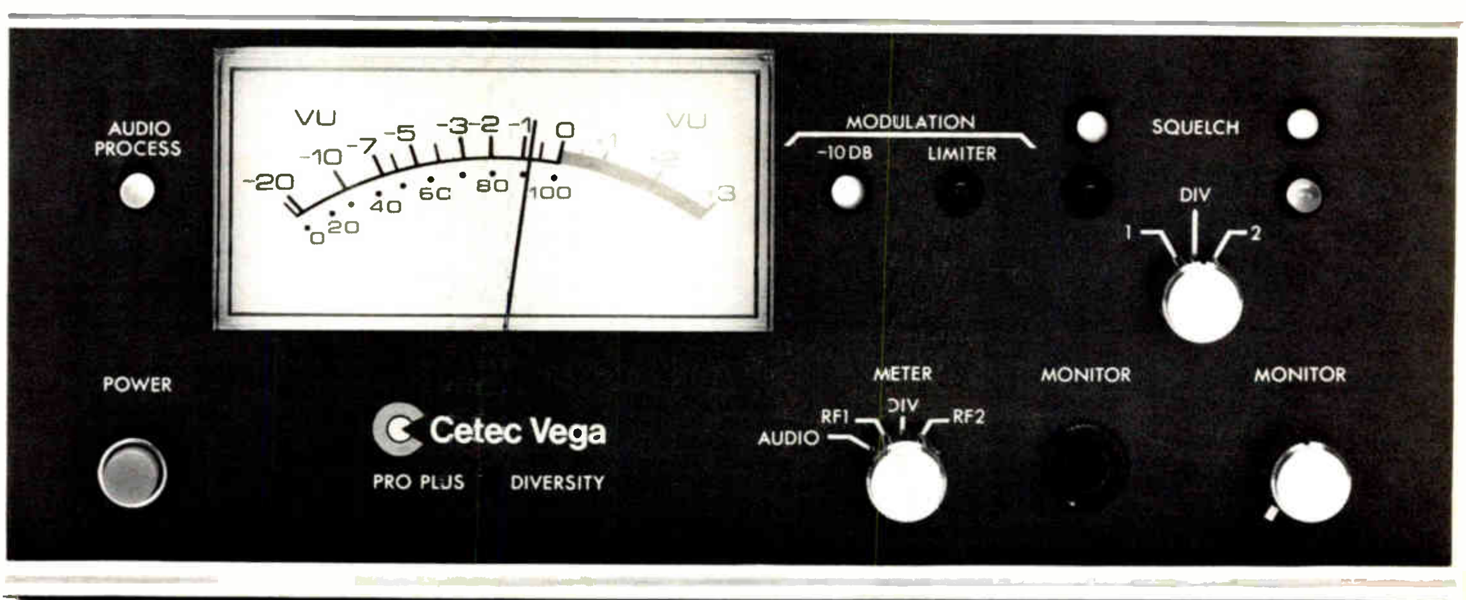
Cetec Vega's top-of-the-line PRO PLUS R-41 and R-42 wireless-microphone receivers have quickly become the worldwide standard of excellence. Overall quality of the PRO PLUS wireless system is equal to wired microphone systems, with respect to dynamic range, signal-to-noise ratio, distortion, etc. *We invite your comparisons.* Check these features of the new, improved PRO PLUS receivers:

- **GaAsFET front end.** Provides the highest achievable sensitivity for maximum system range. Also incorporates a high-performance helical filter.
- **Lowest distortion.** 0.25% maximum, 0.15% typical.
- **Measurably the highest signal-to-noise ratio and widest dynamic range.** Quiet as a wire. With DYNEX II (a new standard in audio processing), SNR is 101 dB (108 dB A-weighted). System dynamic range is 133 dB including transmitter adjustment range, from input for maximum nondistorting gain compression to noise floor.
- **"Infinite gain" receiver.** Improved performance in the critical threshold region, superior handling of multipath conditions, better SNR, and constant receiver audio output level.
- **Professional audio circuits.** Output is adjustable from +20 dBm to -60 dBm in four ranges. Also featured are selectable phasing and 0.2-watt independent headphone amplifier.
- **True dual-receiver diversity.** The R-42 diversity system is the most reliable method to avoid dropouts. The R-41 nondiversity receiver has all of the other features of the R-42.

PRO PLUS wireless-microphone systems achieve the highest performance possible with today's advanced technology.

Write or call for further information and location of your nearest dealer: Cetec Vega, P.O. Box 5348, El Monte, CA 91734. (818) 442-0782.

# The best wireless gets even better.




**Cetec Vega**
*...the professional's wireless*

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# Getting the Most from Contractor's Expo '87

by Marc L. Beningson
Jaffe Acoustics, Inc.

The National Sound and Communications Association Contractor's Expo '87 is being held in New Orleans this April. This is not just another audio related show; it is one that is designed specifically for sound contractors. Many people have mistakenly referred to NSCA as the "National Sound Contractors' Association," which demonstrates the success of the show is appealing to contractors; but that name is incorrect.

The NSCA Contractor's Expo is definitely a show worth attending. (It has become fashionable for many to criticize the Audio Engineering Society for *their* conventions, yet thousands of people gladly attend these shows each year.) The NSCA show has a distinctly different character from AES, which makes it the ideal show for a contractor on a budget. NSCA Contractor's Expo is the only show designed especially for sound contractors and it serves consultants' needs as well. Here are a few pointers on how to get the most out of this (or any other) audio show:

First, schedule everything around the lectures, sessions and workshops. These are the events that can't be duplicated elsewhere, and written papers, magazine articles, and summaries are never as useful as the event itself. A lot of useful, real world information can be obtained by attending and participating in these sessions. Don't just listen and take notes—ask questions, contribute your experiences, and challenge the speakers if you disagree. This makes things more informative and interesting for all the participants.

*Sound & Communications* magazine will be sponsoring two sessions tailored to contractors. One will be chaired by Jesse Klapholz and will deal with speaker array design. This writer will be chairing the second, titled "How to Prepare a Winning Bid," which will provide some insights from consultants and contractors on how to "get the edge" on bidding.

***"Because the NSCA is primarily attended by contractors, you will find few tire kickers, 'goody-grabbers,' and other aimless wanderers taking up space on the exhibition floor."***

Go out and meet other people. One of the main attractions of a convention is that people come from all over the country to attend. This provides you with one of your few opportunities to meet and exchange ideas with contractors, consultants, factory technical personnel and sales representatives, and other industry personalities. Rarely are so many potential resources of information brought together at once, and you should take advantage of this. You might even find that some of the "famous names" of the industry are both friendly and approachable, and would not mind exchanging ideas with you.

*"The NSCA show has a distinctly different character, which makes it the ideal show for a contractor on a budget."*

Because the NSCA Expo is primarily attended by contractors, you will find few "tire kickers," "goody-grabbers," and other aimless wanderers taking up space on the exhibition floor. That means more room for you to take a close look at equipment you might not otherwise have an opportunity to see. Likewise, the factory reps have more quality time to spend with "real" customers like you. Because of this, scheduling your time effectively on the exhibition floor can be more rewarding at NSCA than at many other shows.

Have fun by enjoying the sights, and relax. After all, this is one of the few times you get out of the office or the shop, and you should have a good time. Relaxation time can also be quality time if you get to know other people from the industry by relaxing with them. Having a few drinks or going sightseeing with these people can establish good working relationships that will certainly be rewarding. At the very least, you might make a new friend. It's easier to do business with someone you know well.

*"Don't just listen and take notes—ask questions, contribute your experience, and challenge the speakers if you disagree."*

# WIRELESS MICROPHONES

PART II

# Diversity Antenna Systems

by Bill Sien & Steve Barbar

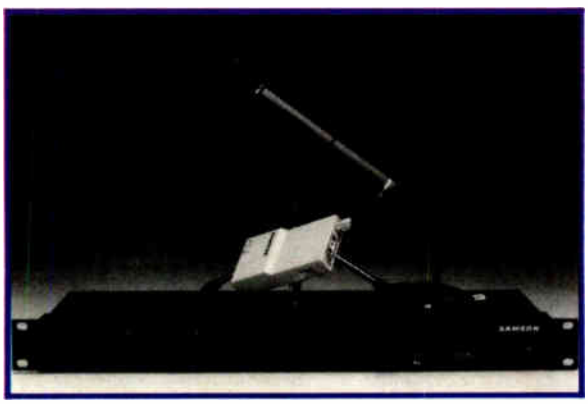
Part 1 of this article focused on the general field of wireless microphones and improvements in the technology. We also mentioned the situations when more than just a simple antenna would be required for clean wireless microphone operation. In this installment we will take an in-depth look at the various antenna systems available to the professional user of wireless microphones.

### Diversity

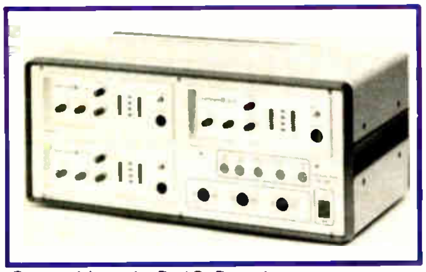
Diversity systems take advantage of the fact that multipath zones are small. If the diversity antennas are spaced a half-wave apart and/or are oriented in different planes, each antenna may briefly be in a multipath zone. However, the statistical probability is extremely small that it will occur at both antennas simultaneously. If circuitry is provided to continuously and instantaneously select the antenna which has the best signal, or to combine the outputs of such antennas, the probability of a multipath noise-up—or worse; a dropout—can be reduced by the same factor. Extensive testing and field use has shown that this improvement is realized in practice.

There are currently three types of wireless microphone diversity systems on the market: *dual-receiver* or "true" diversity; *multiple-antenna* or "antenna" diversity; and *phase diversity*. True diversity employs two complete receivers tuned to the same frequency along with control logic after the audio amplifier to instantaneously select the audio from the receiver channel having the highest RF input signal—and therefore, the best S/N (signal-to-noise-ratio). Using this approach, the audio is always developed from the best RF signal currently available, and the appropriate receiver audio is fed to the audio output. In addition, true diversity has the advantage of providing redundancy. There is only one disadvantage to the true diversity design, albeit an important one-cost. A multiple-channel receiver is always significantly more expensive than a single-channel design, due both to the second channel and to the extra cost of the detection and switching circuitry itself. In all other respects it is superior.

*Antenna diversity* requires three antennas rather than the two needed for the other systems because of the summation of phase information in addition to amplitude information that is fed into a summing network. The output of the summing network is the vector sum of the signal present at the individual antennas. The probability of simultaneous multipath nulls at two antennas is no higher than for true diversity. However, the probability of the relative phase between the two signals approaching 180-degrees resulting in an output null even under strong signal conditions is much higher. The addition of a third antenna largely avoids this problem



Samson's Concert TD Series



Cetec Vega's R-42 Receiver.

by adding another independent term to the vector sum. An isolation amplifier for each antenna is necessary in order for the individual antenna to be electrically independent. If the output of the three antennas is merely passively summed, an antenna array is synthesized. Such an array has a definable electrical center and behaves essentially as a single antenna. In this case, no immunity from multipath nulls is obtained.

The biggest advantage to antenna diversity is cost. A simple diversity adapter, which is economical to design and manufacture, is provided. This approach does have the additional advantage to the user of permitting the decision to purchase the antenna diversity hardware to be deferred to some future time.

There are, however, several serious disadvantages. The first, the summing network described above, becomes the receiver front-end. Since practical considerations dictate that the amplifier be fairly broadband, intermodulation, overload, and desensitization are frequently troublesome. The front-end of the actual receiver, however good, cannot correct these problems. Another problem is that as the performer moves, a random phase summation occurs in the antenna diversity system. The output to the receiver will exhibit rapid changes in both phase and amplitude which are audible. Furthermore, the output of the summing network is almost never as high as that of the best input channel, since the summing network is summing both amplitude and phase. With the exception of a few special cases, such as when the outputs of both of the other two channels is zero or when the outputs of all three antennas are exactly in phase, the output of the summing network will be less than that of the best channel. Under weak-signal conditions, the audio S/N can be expected to be generally lower than for a true diversity system.

Telex's FMR-2 receiver with WHM-500 handheld microphone/transmitter.

A practical drawback to antenna diversity, which may not always be a significant factor, is that three antennas take more time and effort to set up than two. In addition, to be effective, somewhat more antenna-to-antenna separation is needed than is necessary for true diversity.

*Phase-switching* diversity is a hybrid between true diversity and antenna diversity techniques and relies upon summation of the outputs of a pair of appropriately spaced antennas. With widely spaced antennas, the problem of the vector sum of two strong signals approaching zero is very real. If the antenna spacing is small, the summation will result in an array formation, and the two antennas will behave as a single antenna. Phase diversity approaches these problems by providing electronic circuitry to switch the phase of the signal from one antenna when the control circuitry in the receiver detects a drop in signal level. Assuming that the drop in signal level was due to the relative phase approaching 180-degrees or a multipath null at the array center point (and not propagation effects), the resulting new vector sum will be safely removed from the region of the null in the original vector sum. On the other hand, if the control circuitry is triggered by a signal-level drop due to propagation effects, the switch may actually make matters worse by switching to the 180-degree relative phase region or by displacing the array center to a multipath null. Therefore, the control circuitry must be able to switch back to the original condition if the signal level does not improve. An additional problem that must be accounted for in a properly designed system is that the phase switching produces a phase discontinuity in the RF signal. This can appear at the FM demodulator output in the form of a transient waveform.

Shure's W20 R Receiver, W25DR Diversity Receiver, WL83 Lavalier Microphone and W10BT Body-Pack transmitter.

Swintek's dB-S System.

As with antenna diversity, phase diversity has the single biggest advantage of being less expensive than true diversity. However, unlike antenna diversity, a special receiver design is necessary. While the use of some type of audio processing is helpful in both true diversity and antenna diversity, it is essential for phase diversity in order to mask the audio transients resulting from rapid RF phase excursions and to overcome the sudden drops in output S/N caused by inappropriate antenna phase switching. That is, the signal level drop may be due to the performer walking behind absorptive material, and not due to a multipath null or 180-degree relative antenna phase. In this case, a switch may result in the creation of a null condition on top of a low-signal condition. In addition, the switch may result in the displacement of the antenna electrical center along the line of the multipath null or along the line of the 180-degree phase difference, in which case no improvement is realized. The one big concern

*(continued on page 22)*

Sennheiser's EM 1036 System.

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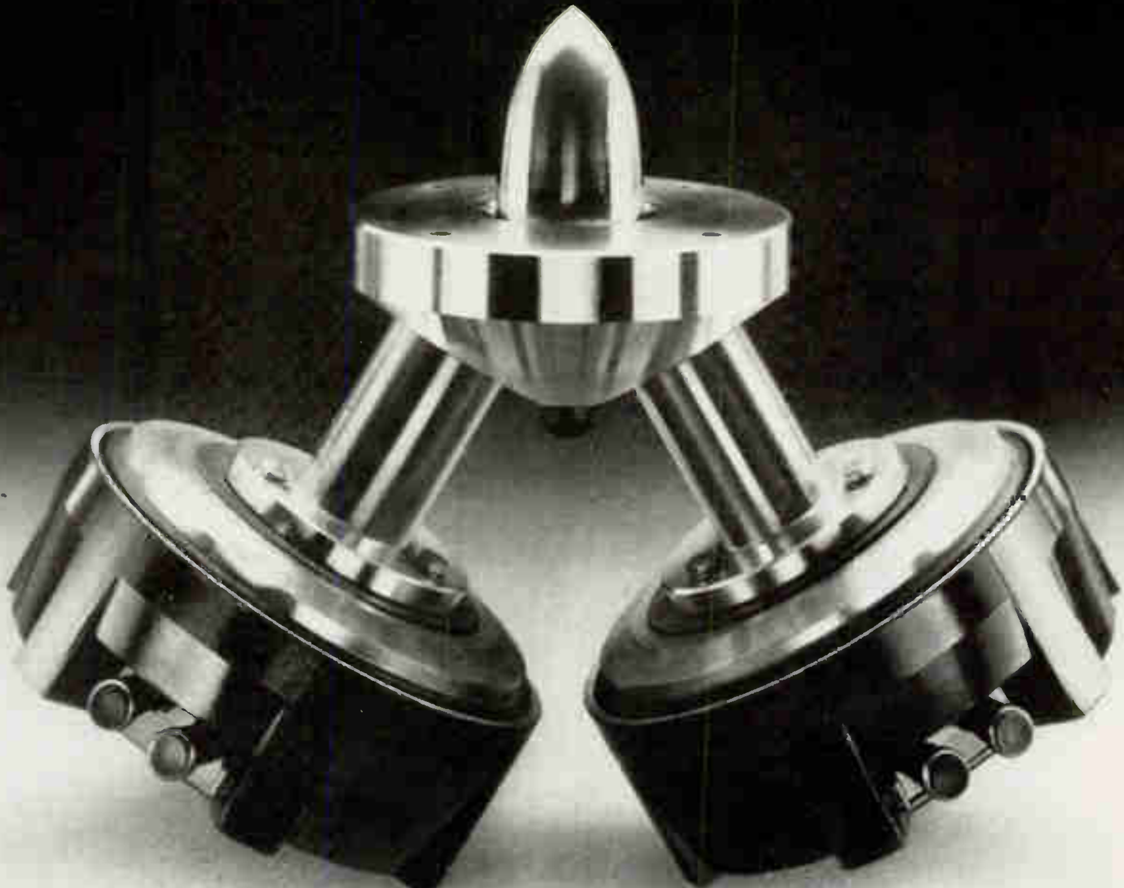

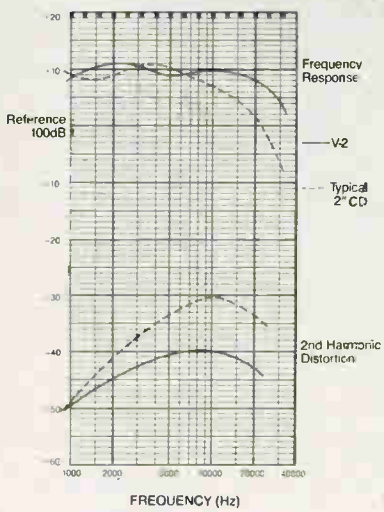


Curves have been smoothed for purposes of comparison.

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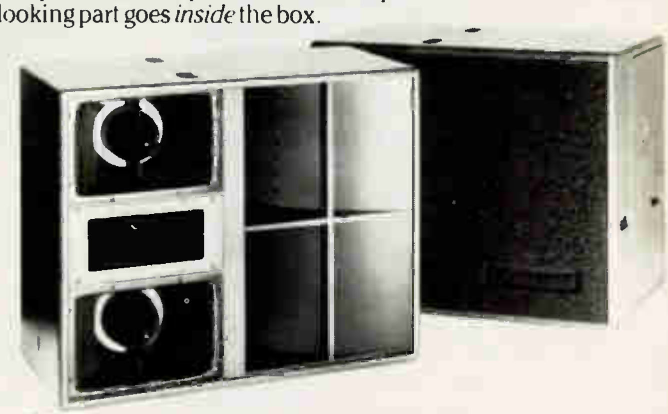

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# Wireless Beams Down "Sound Cloud"

**by Royce Krilanovich**
Nady Systems

This was the scene at Battery Park near New York's Hudson River on the evening of September 13, 1986. Throngs of people young and old—as diverse as New York itself—gazing at the transparent pyramid suspended in the darkening sky, or somewhere looking toward the river, they marveled at the barges bristling with scaffolding, antennas and lights. A festival atmosphere prevailed as an elegant white boat drifted into position. Two helicopters buzzed overhead. All awaited a cue from the pyramid.

Suddenly the sky gleamed with lasers and fireworks, and the earth shook with the music that rolled out of the sky in a sound cloud, enveloping the crowd. Isao Tomita had begun his concert, "Back To The Earth."

Tomita, the venerated pioneer of electronic music, was making a rare public appearance. Tomita conceived as a musical celebration of the cosmos and the earth. A tremendous undertaking, it was on the cutting edge of electronic music technology and a tribute to the concert sound and lighting contractors—ShowCo of Dallas and WesTec Audio of New York—who worked so hard to mount the show.

Tomita's intention was to center the outstanding effects of the performance around a "Pyramid Capsule," jointly representing the beauty of light and sound to create a "total sound zone." Inside the capsule Tomita would control a complex electronic system, commanding powerful lasers and sending forth the audio mix to giant speaker stacks on land, on barges, and in the air. Besides the music he would be playing, Tomita would also be wedging in the audio from performers on barges.

These artists included the Choirs of the Cathedral of Saint John the Divine; from Japan, Goro Yamaguchi playing Shakuhachi flute and Marido Senju on violin solos; from the People's Republic of China, Chin Lin played chinese lute; and from the USSR, pianist Nikolai Demidenko pleased the crowd with Gershwin's "Rhapsody in Blue."

Steve Winawer of Westec Audio provided the wireless link that would bring the performance together. Nady Wireless Systems had been specified by the principal contractor, ShowCo. Winawer called Nady's sales manager, Pete Kalmen, and presented his problem: for this performance, WesTec would need 11 separate channels of wireless. Each had to function in the RF-polluted environment of New York City—at distances of up to 1,000 feet. Kalmen, in turn, laid out the challenge to Nady's engineering staff, and there followed weeks of daily conferences between WesTec and Nady.

Tough requirements in RF gear design were not new to Nady engineers. (In fact, the company's founder and president, John Nady, received a U.S. patent for his pioneering work perfecting wireless microphone systems.) Nady's group felt that the number of systems needed could be provided—even considering that many of the roughly 20 FCC-assigned VHF high-band channels might not be available. Very sensitive front-end filtering is the key to success when using multiple wireless systems in close proximity. Nady's newest generation filtering circuitry had proven itself in the field under similar circumstances.

A far tougher demand was the stated range requirement. The plan called for six outgoing channels of wireless to carry audio mixes to speaker barges and performer's barges. One outgoing channel was to carry a click track to performers, and another would transmit a mix to a helicopter with amp rack and horns flying overhead. In addition, two incoming wireless channels would be used so audio from the chorus and violin soloist could be mixed and sent out to the speaker stacks on land and barges (see diagram). The barges would be up to 1,000 feet away from Tomita's control pyramid—the wireless systems employed had to have this much range and then some for reliable transmission.

The WesTec and ShowCo people had expected Nady to meet the range requirement—if indeed it could be met—by the use of power amplifiers. This somewhat conventional (though expensive and illegal) answer was considered by the Nady engineering staff, but rejected. Instead, they decided to pursue an idea that came up during a brain-storming session, combining RF and antenna technology.

Rather than up transmission power, Nady experimented with frequency matched antennas known as "Yagi" antennas. The principle behind these antennas is that they are selective in frequency and selective in direction, effectively increasing the transmission strength and the sensitivity of the receiver with no added output power. They have the added advantage of not adding to the site's considerable RF pollution with more power. And, because the signal and reception with a Yagi antenna is so focused, wireless systems employing them are rendered largely immune to RF on neighboring channels. It so happened that one of the Nady engineers on the project had designed Yagi antennas for long distance television broadcasts, so the necessary expertise was already in place.

Yagi transmit/receive antennas near control pyramid.

A test unit was built. On a balmy summer afternoon, Pete Kalmen and two engineers set up the system to transmit across a local lake.

It worked. With stock Nady 701 VHF transmitter and receiver, they coaxed a mile's range out of the system—twice as much as they had need for the Tomita installation. Kalmen was on the phone early the next day with the news. ShowCo's Gary Epstein wanted to see—and hear—for himself. Audio quality was absolutely critical to this show, and he wanted to make sure that the extraordinary antennas sent and received every nuance. At Kalmen's lakeside test bed, Epstein observed as the system was put through its paces, and passed every test with flying colors.

Steve Winawer spent a number of Saturday afternoons in New York City with a spectrum analyzer determining what channels were likely to be clear. This information was passed to Nady, who built the wireless systems on these channels with Yagi antennas to match. In all, Nady would build 30 custom Yagi antennas, and provide 12 wireless systems for the Tomita performance.

In the words of Winawer, the event was "super." As harried and tired as he was from the hard work of set up, he enjoyed the show.

Isao Tomita had expressed his wishes for the event in this way, "Through this sound and light experience, I hope you will reflect upon the wonders of the universe, and Mother Earth...and ultimately the experience will serve to enhance better understanding, love, and peace in the world."

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## WIRELESS MICROPHONES

*(continued from page 22)*

is that in spite of the diversity circuitry, there exists a somewhat greater probability of a rare total dropout than with the other two systems discussed.

AKG's SR 185 wireless microphone system.

As with antenna diversity, effective operation requires somewhat greater antenna spacing for phase diversity than is needed for a true diversity system. The average improvement in multipath null depth is approximately the same as for antenna diversity, but incorrect phase switching by the control circuitry can cause a momentary dropout that the antenna diversity system would not experience. Nevertheless, performance is significantly better than with an equivalent non-diversity system, particularly when the average signal level is high.

Diversity systems are well worth investigating when considering wireless microphone technology. Understanding how the various antenna systems operate will enable the system designer to properly select the appropriate system for a given installation. Also, understanding how these antenna systems interface with the environment and application will provide for easier and smoother installations and operation.

---


**Bill Sien, Cetec Vega dealer, is a principal of Systems Wireless, Ltd. and has a B.S. form the University of Maryland. A former radar specialist with the military, he has authored several articles in various trade journals.**

**Steve Barbar has done extensive course work at the University of Maryland while working in their A/V Department. He has over seven years working experience in the field of audio doing studio recording and live concert mixing.**

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# SPEECH SYNTHESIS AND VOICE RECOGNITION

by Jesse Klapholz

The field of Speech Synthesis and Voice Recognition has been under development since the beginning of the science of communications. Even before the development of the telephone, experimentation in speech synthesis had begun as part of the study of speech sounds. After the introduction of computer technology around World War II, and with the intensified need for rapid and accurate communications by industry and the military, speaker recognition studies were extended from speech/ear/brain channels to speech/mic/computer channels. The fields of speech synthesis and computer voice recognition are rapidly becoming a part of the field of sound and communications.

### History of Speech Synthesis

While the first speech synthesizers were mechanical, these circa-1800s machines are of curious interest as to how they actually worked. However, the principles those early machines used to mimic speech sounds illustrate the understanding of speech acoustics by their inventors. Scientific explanations of speech production actually date back to the ancient Greeks and Romans. They had classified speech sounds into vowels, semi-vowels (*i.e.*, voiced consonants), and consonants (*i.e.*, voiceless consonants).

The period that followed leading up to the 17th century, was one that preoccupied most speech research scholars with the attempts of constructing artificial talking automations. There are many stories and fables too long to tell here. Some machines were later reproduced, proving in some fashion that the ancients did indeed have some modicum of technology developed. Some were fakes, using long tubes into which a remote talker would utter his words; others even included bogus machinery used to hide a dwarf!

The first scientific record of speech synthesis was during the early years of the French Academy of Sciences which was founded in 1666. Abbe Mical of Paris is credited with the construction of the first artificial talking machine some time between 1750 and 1780. Shortly thereafter in Russia, Christian Gottlieb Kratzenstein constructed a set of resonators similar in form and size to the human mouth. Through these resonators he passed an interrupted air stream produced by an organ-type reed. Wolfgang von Kempelen of Vienna devised an artificial vowel machine in 1791. Although at the time he was not recognized for his invention, Sir Charles Wheatstone built an improved version with far reaching influences, perhaps more so than is commonly known.

During Alexander Graham Bell's boyhood in Edinburgh, Scotland, Bell had an opportunity to see Wheatstone's von Kempelen talking machine. Bell was determined to build a talking machine of his own. With help from his brother he succeeded in doing so. He claimed that his machine could enunciate vowels and nasals, sufficient to yield a few simple utterances. The rest is speculative history.

Hermann von Helmholtz experimented with an array of harmonically assembled and electrically driven tuning forks which produced several vowel sounds. In 1911, Marage (no relation to the Mirage synthesizer), a Parisian scientist, built an electrically driven set of air bellows that emulated the lungs and, by using artificial larynxes and mouth and

Courtesy Paramount Pictures

In *Star Trek IV The Voyage Home*, Scotty attempts to communicate with a computer using a mouse. The engineer from the future doesn't realize that computers in the 1980's respond to commands given via the keyboard.

nasal passages, was able to produce a limited set of vowel sounds. At about the same time, Dayton C. Miller, at Case University, used sets of organ pipes to produce limited sets of vowel sounds.

## Electrical Talking Machines

The engineering approach to speech synthesis has historically used the technique of modeling the frequency-transmission characteristics of the vocal tract, producing its electrical analog. Typically, speech synthesizers using this analog technique do so from a terminal point of view, *i.e.*, at the end of the series of components in the model. These devices have been called terminal-analog, or *formant*, synthesizers. The term analog used here does not specify analog or digital circuitry, only that an electrical circuit is used to model our speech system.

Acoustic theory shows that the transmission characteristic of the vocal tract can be closely approximated by a cascade of uncoupled resonators and antiresonators whose bandwidths and center frequencies may be independently controlled. Therefore, the history of electrical speech synthesizers begins with the development of the terminal-analog family and shows the progress made in devising better resonant circuits and methods to more accurately control them.

John Q. Stewart constructed the first electrical analog of the vocal organs in 1922 at AT&T. He devised a network that was manually controlled via variable capacitors that was capable of producing a restricted class of vowel-like sounds. Similar apparatus was demonstrated by Dr. Harvey Fletcher of Bell Labs at the New York Electrical Society in February 1924 when he produced a limited group of sounds including some vowels and the words "mama" and "papa."

Subsequently, analyzers and synthesizers became an integral part of the research work in speech and musical acoustics. In the 1930's work at Bell Labs and UCLA led to two independent reports by Fletcher and Knudsen which both stated the necessity of an electronic synthesizer capable of combining a large number of harmonic partials. In 1940 Bell Labs built a 100-element harmonic tone synthesizer. Its sound source was 100 separate harmonically-tuned sine waves recorded on a rotating magnetic drum.

The name Homer Dudley may not be well known to many in the audio business, but Dudley worked at Bell Labs through the beginnings of electronic speech synthesis and was responsible for the Vocoder (Voice CODER) in 1936, and shortly thereafter the VODER (Voice Operation Demonstrator). The VODER required a year's training to "play" it and was first demonstrated at the Franklin Institute in Philadelphia in 1939. The VODER was subsequently demonstrated to the public at the World's Fairs in New York and San Francisco in 1939-40. This was the public's first exposure to electronic speech synthesis.

## Sound Spectrograph

During World War II, R.K. Potter at Bell Labs was working on a new frequency analyzer that gave a continuous real-time analysis of speech. It was called "Visible Speech." In his book, Potter describes his device: "One speaks into a microphone and the oscillations of his speech are then passed through 12 electrical filters....When amplified, each filtered set of oscillations lights a tiny grain-of-wheat lamp; there are twelve lamps, arranged vertically. The fundamental tone of the speech lights one lamp, the second harmonic another further up, and so on. The lamps that light in response to the speaker indicate the frequencies present in his speech.... The result is a characteristic pattern for each vowel and consonant, defined by lines of varying frequency and duration."

Work continued on at Bell Labs in identifying the intricacies of human speech, and in 1950 H. K. Dunn had devised an electrical vocal tract based on the analog of its human counterpart. The passive circuit, which modified a vibratory energy source, duplicated the "delays" in the vocal tract with a number of low-pass filter sections. Two other coils, with variable inductance, were used to simulate the "inductance" of the tongue and lips.

By the beginning of the 1950s, workers at Bell Labs had already

An inspector from Austin-Rover is verbally inputting data to a central computer while performing quality checks.

Worker at West Point Pepperell's textile plant puts information into a computer by speaking into Votan equipment.

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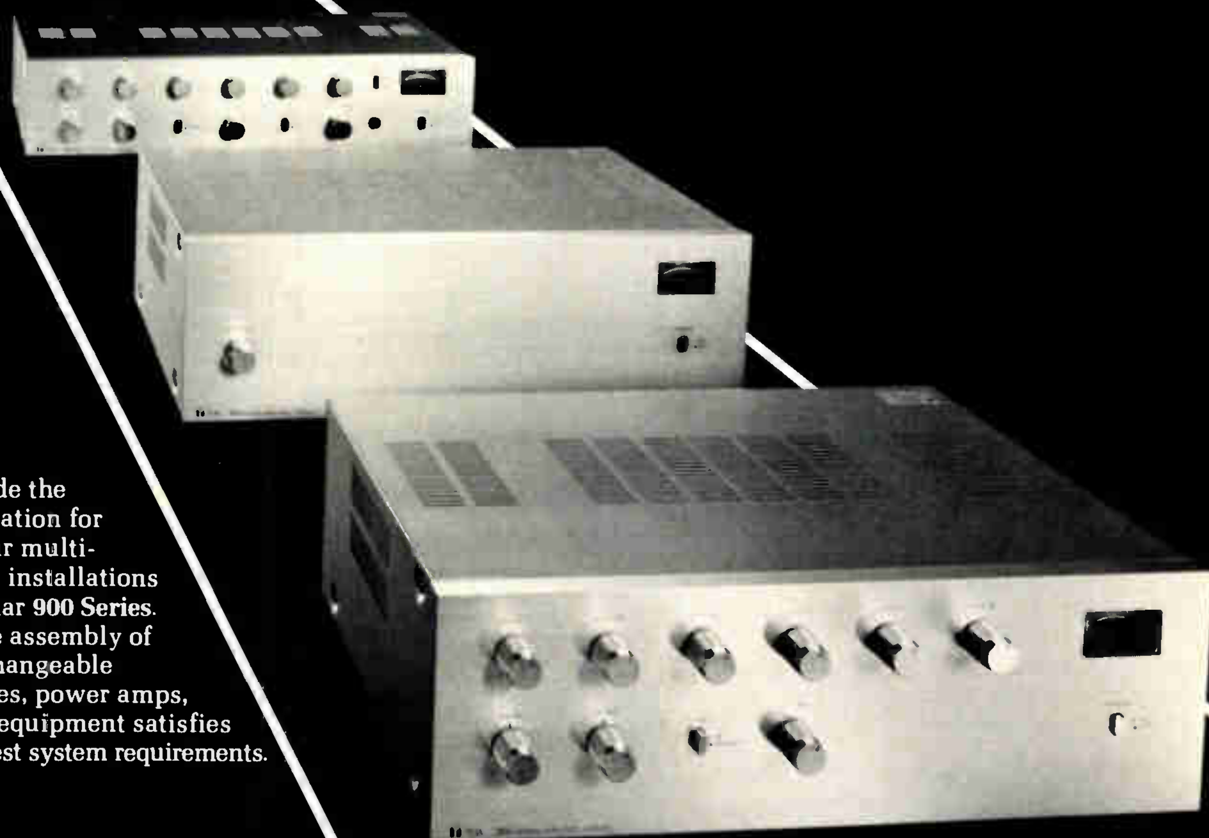

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started to apply what they had learned about speech production to speech recognition. Their first project was AUDREY (Automatic Digit Recognizer). AUDREY was capable of recognizing ten basic phonetic digit sounds. This device was the beginning of research into the electronic recognition of speech sounds, or phonemes, which would ultimately lead to a vocabulary-based word recognition system.

In the early 1950's a synthesizer program was under development at RCA under the direction of Harry F. Olson and Herbert Belar which resulted in the RCA Electronic Music Synthesizer. The RCA synthesizer was programmable in its control parameters over a tone's frequency, intensity, growth, duration, decay, portamento, timbre, and vibrato. A punched paper roll contained all the information of a performance and could change any parameter. The RCA synthesizer was initially used as a musical instrument and musical composition analysis tool. However, this technology was later applied to speech analysis and voice synthesis; and by 1954 a few sentences had been synthesized and recorded.

By 1956 a "Phonetic Typewriter" had been developed at RCA which used techniques similar to Visible Speech to form a *syllable recognizer.* The device could recognize succinctly spoken words and type them with 98 percent accuracy when they were coded for a single person. Later this project was expanded and at a meeting of the Acoustical Society of America in 1962, Olson, Belar, and Ricardo de Sobrino demonstrated a speech processing system that consisted of a speech analyzer, translator, typer, and synthesizer. The system used a magnetic syllable storage drum that was capable of synthesized speech in English, French, and Spanish.

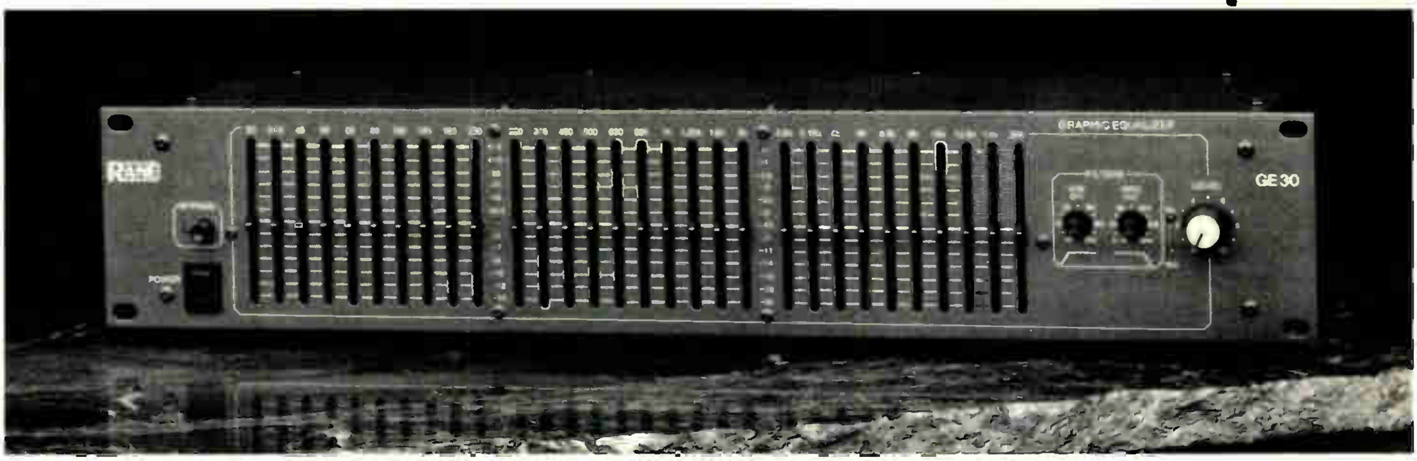
While work was going on at RCA using analog techniques, Bell Labs was starting work on digital synthesis. In 1957, Max Mathews first demonstrated his work on digital speech encoding. The IBM 704 computer used in his work enabled him to study encodings in several months that would have otherwise taken years to develop

*(continued on page 48)*

# IN-TER-PO-LATE

in-ter-po-late (in tur′ pə lāt) vt. [L. <inter-between + polire, to polish>] 2. to insert between or among others.

## INTRODUCING THE GE 30 INTERPOLATING CONSTANT-Q GRAPHIC EQUALIZER.



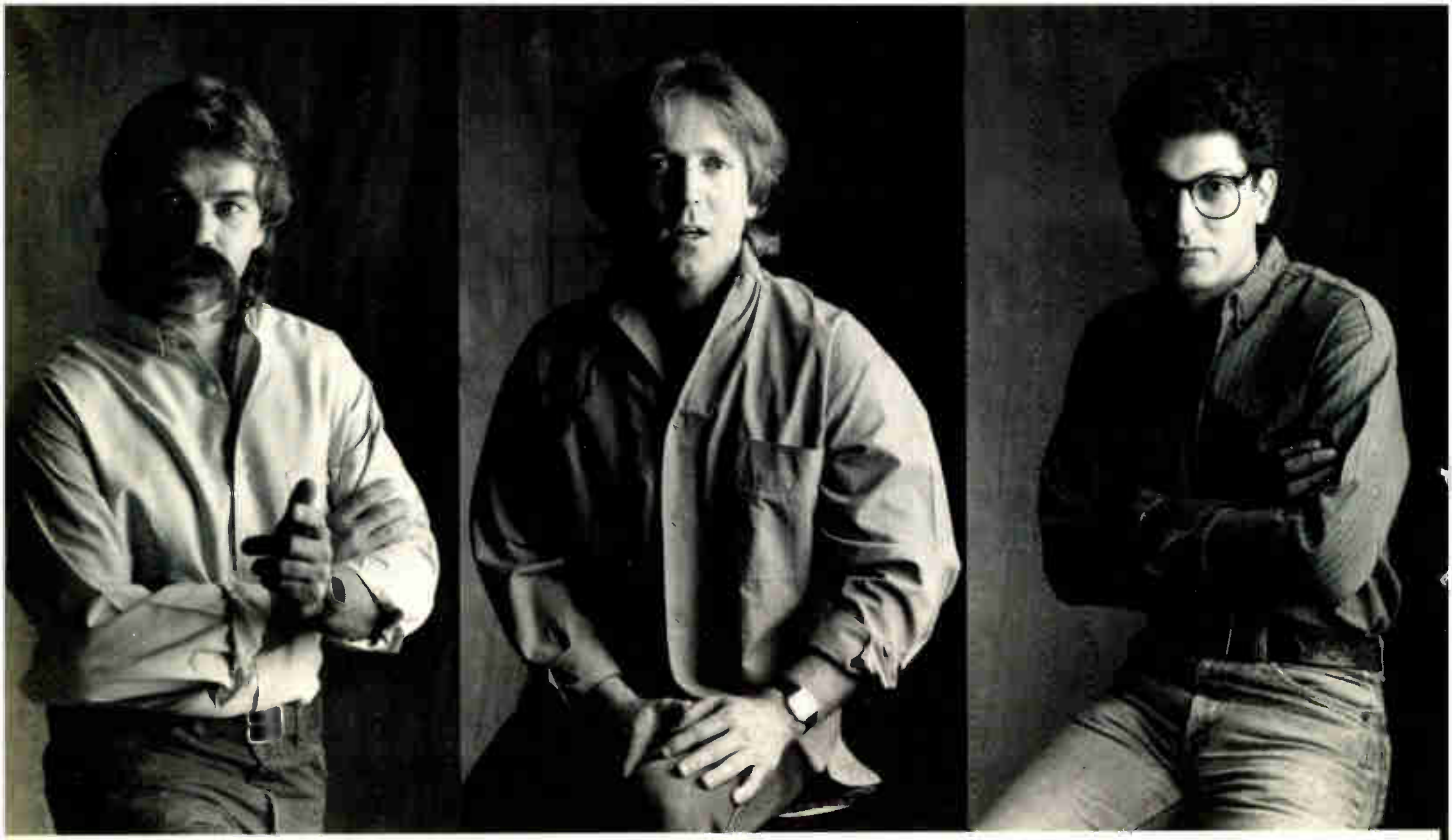

Rane has taken the graphic equalizer format to yet another new level of performance capability.

In response to valuable input from leading acoustical engineers, Rane has redesigned the GE 30 to enable Constant-Q bandpass response ON OR ANYWHERE IN BETWEEN THE ISO 1/3 OCTAVE CENTERS, for precise normalization of acoustical nodes, with minimal adverse effect on surrounding frequency response.

Through the use of dual alternating summing amplifiers and precise computer-optimized Constant-Q filter characteristics, the combined use of two adjacent ISO filters on the GE 30 results in a well-behaved "interpolated" bandpass response BETWEEN THE ISO CENTERS with only slight degradation in Q and a minimal increase in boost/cut. The exact center frequency of the combined filters can be "tuned" in either direction by simply adjusting the level of one of the two sliders, shifting the "nose" of the filter's response towards the slider with greater boost/cut. This allows precise alignment of the filter center frequency with that of the acoustical node for optimum end results.

The accuracy and sophistication of today's computerized acoustical analysis systems has put increasing pressure on signal processors to solve a myriad of acoustical phenomena that are now precisely identifiable. By combining interpolating capability with the already proven Constant-Q performance, the GE 30 provides an unprecedented degree of control with consistent precision, yielding optimum sound quality in complex acoustical environments with a minimum of effort and hardware.

RANE CORPORATION



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South Coast Sound
Live Sound Engineer

"I Got A Cleaner Sound With More Punch And Tighter Bass."

Danny Watson
Promise Productions
Producer/Keyboardist

"We Can Customize The Sound System For Each Job."

Miles Kapper
Sight-On-Scene Staging
AV Staging Specialist

Three professionals whose careers depend on getting excellent sound — and though each has a different set of criteria, all agree on the benefits of QSC's Octal Accessories. QSC has pioneered the development of high power, low-profile amplifiers of unquestionable fidelity and reliability. And with the new Octal Accessories we've also pioneered versatility. QSC's unique Octal socket [which is standard on all Series One and Series Three amps] accommodates a whole line of passive and active Octal Input Modules — from input transformers to electronic crossovers and power limiters — audio accessories that enhance and customize the performance of each QSC amplifier, quickly and easily, and at a cost far less than buying rack-mounted devices. And when requirements change, a different QSC Octal module can meet that need. In this way QSC is making obsolescence a thing of the past. QSC Octal Products are space-savers too: each module attaches at the rear panel, out of the way without adding rack space. All Octal Accessories perform to the same high standards that distinguish QSC amplifiers, combining meticulous design considerations with real-world durability. Octal Modules include: XH-1 and XL-1 Crossovers, PL-1 Compressor/Limiter, UF-1 Universal Active Filter, T-1 Input Transformer, A-1 Octal Attenuator, and AT-1 Octal Attenuator with Input Transformer. So no matter where your sound matters — on stage, in the studio or in a custom designed installation, QSC provides your sound system with an invaluable lasting benefit: versatility. For more information contact: QSC Audio Products, 1926 Placentia Avenue, Costa Mesa, CA 92627 [714] 645 2540.

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Patrick Quilter
Vice President/Engineering,
QSC Audio.

Barry Andrews
President.
QSC Audio.

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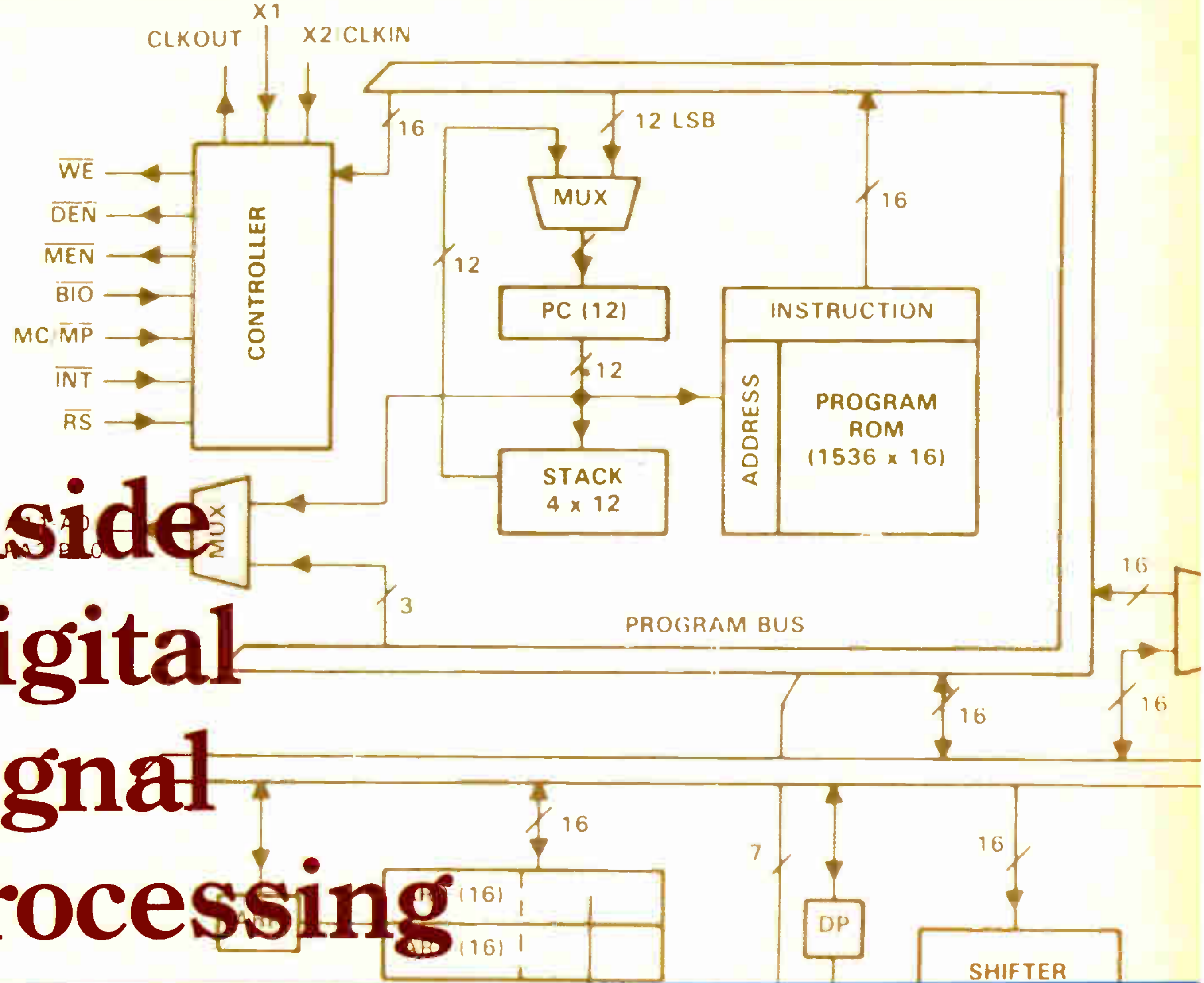



# Inside Digital Signal Processing

## THE DSP CHIP

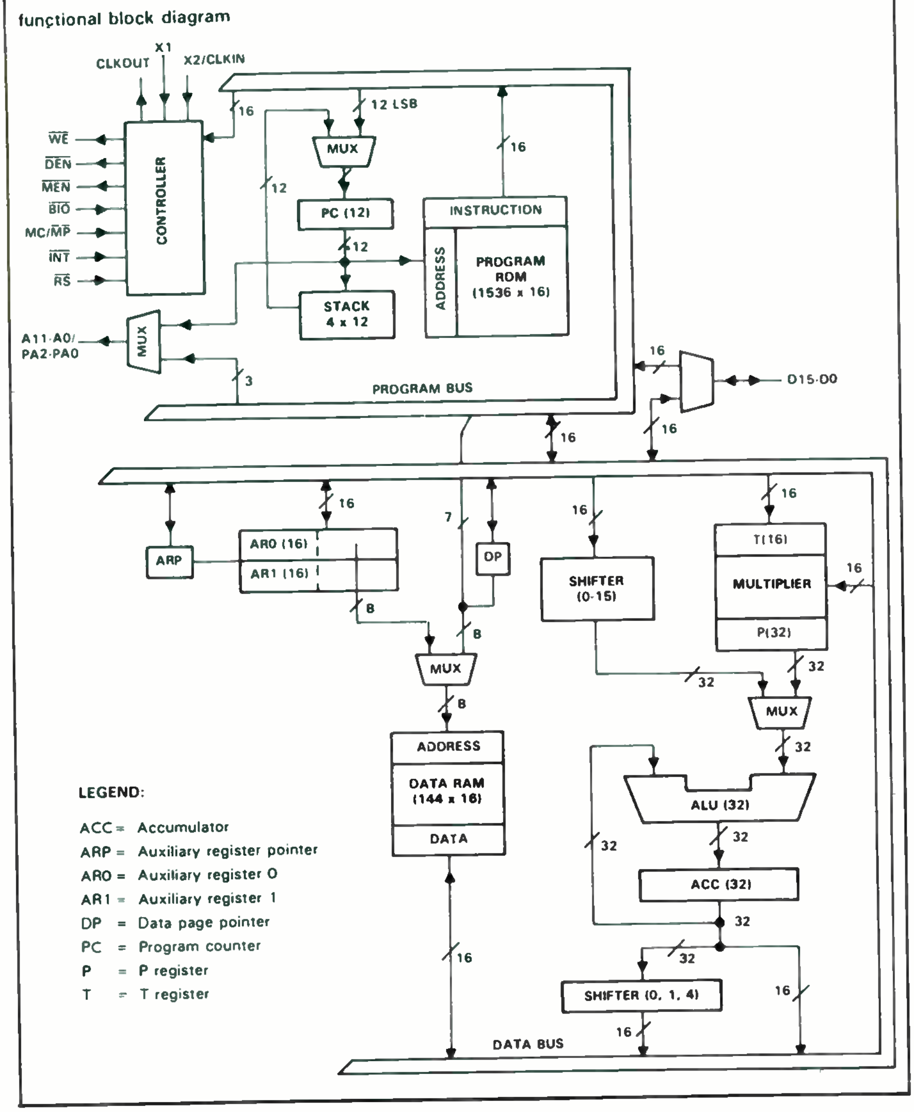
By Michael L Beigel

### INTRODUCTION

The parable of the pocket calculator is one of my favorites, and I use it every time I want to call attention to an emerging technology breakthrough. It is actually a mental exercise to gain perspective.

The first thing you do is remember when desktop electronic calculators first came out. They were big and heavy and cost a couple of thousand dollars. The next thing you do is take a look at a ten dollar "credit card" calculator and try to absorb the extent of the change in size, price, and capability of digital technology over only a few years. The last thing you do is to hold up the credit card calculator (or your LCD watch

**Figure 1:** System Diagram for the Texas Instruments TMS32010




if it is more convenient) next to any big, cumbersome, expensive piece of equipment you routinely deal with, and let your imagination go...

Digital Signal Processing is now completing the transformation from laboratory curiosity to commercial fact. Right now it's about at the "big desktop calculator" stage. When Texas Instruments introduced the first DSP chip (the TMS32010) in 1983, it sold for about $200. Now the same chip family sells for about $15, and continues to decrease in price.

T.I. has already introduced a successor (the TMS32020), and numerous other companies are entering the market with similar but distinct DSP chips. Analog Devices, Fujitsu, Motorola, NEC, Weitek, and others have introduced DSP devices.

Yamaha has introduced a completely digital mixing board using custom DSP chips. Digital reverb systems are dropping in price as they increase in quality.

Can ordinary mortals hope to understand the inner workings of digital signal processors? Can ordinary writers hope to explain the inner workings of digital signal processors? Let's find out.

In "Digital Signal Processing" (*Sound and Communications*, December, 1986), we discussed the fundamental aspects and issues in audio which justify the use of specialized digital signal processing chips.

We'll now go on to explain the basic concepts of these special microcomputers which are architecturally optimized for real-time processing tasks which were once handled exclusively by analog components.

As an example of DSP Device, let's take a look inside the first commercially introduced DSP chip, Texas Instruments' TMS 32010.

### Harvard Architecture

Figure 1 depicts the system diagram for the Texas Instruments TMS32010 processor, which is basically a "Harvard Architecture" device (yes, the concept originated at Harvard University).

The main distinguishing feature of the so-called "Harvard Architecture" processors is the use of two separate busses and memory banks for program and data.

While a conventional microprocessor uses a single "bus" and memory for all program and data communication between the computing elements and the memory, the "Harvard Architecture" uses one bus and memory for interfacing to the program and another bus and memory for interfacing to the data to be processed. The use of two busses allows simultaneous interaction of the arithmetic processor with the instructions that tell it what to do and with the data stream it is "doing" it to.

Instead of having to get an instruction and then get the data to operate on, the instruction and data streams flow simultaneously. The processing system is thus "opened up" for faster computation and data flow.

### Data Bus Subsystems

The addition of specialized high-speed computation sections is another distinguishing feature of the DSP device. There are four basic arithmetic elements: the ALU, the Accumulator, the Multiplier and the Shifters.

The *Multiplier* is probably the most significant of these sections in distinguishing a DSP processor from a con-

Lexicon Opus Audio Production System Workstation.

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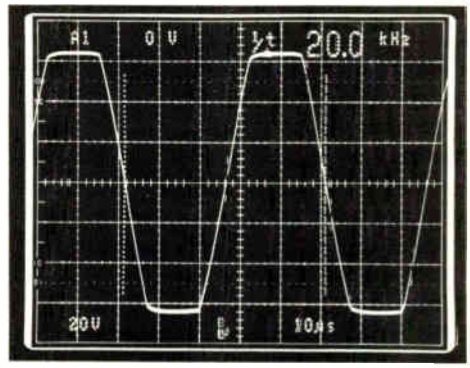
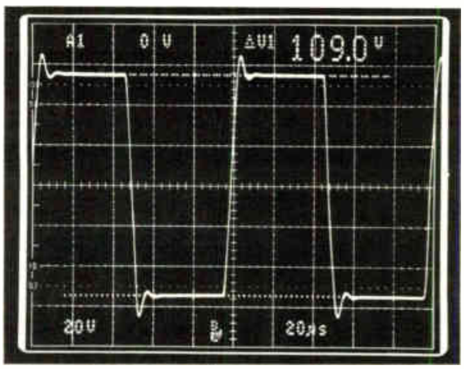



The WP-9920 exhibits gentle overload characteristics even when overdriven 3dB into 4Ω Load at 20 kHz (643 Watts).

characteristics dramatically improve apparent headroom. To learn more, see your RAMSA dealer, or write to 6550 Katella Avenue, Cypress, CA 90630.




The WP-9220 exhibits minimum ringing even with extremely reactive loads. (8Ω, 1µF; 10 kHz square wave input.)

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ventional microprocessor. In a standard microprocessor, the multiplication function is implemented as a lengthy series of arithmetic addition and shifting operations—actually a little "program" itself. The execution of the multiplication function consumes many "clock cycles," and therefore quite a bit of time.

The multiplier section in a DSP processor is an additional piece of hardware that multiplies two numbers in a single clock cycle, approximately 50 to 70 times faster than the conventional method. This feature accounts for the greatest speed improvement

Yamaha DMP7

factor of the DSP chip versus the conventional chip.

The multiplier operates on two 16 bit inputs and delivers a 32 bit output. This highlights another architectural aspect of the DSP processor. While interfacing to the "outside world" via a 16 bit data path (the current industry standard for audio), the internal arithmetic in the processor is performed in


ART Proverb

a 32 bit binary format. The 32 bit internal computation system is necessary for adequate accuracy in processing 16 bit audio information.

The *Arithmetic Logic Unit* (ALU) performs addition, subtraction, and logical functions on the incoming 32 bit data. In conventional microcomputers, the ALU is usually the only mathematical processor in the system.

The *accumulator* stores the results of operations performed by the Multiplier and the ALU, and is also often an input to the ALU.

The *shifters* provide the ability to easily change the scale or relative magnitude of the incoming and outgoing data streams. This allows the processing elements to operate on the data in the optimum range to provide both high precision and freedom from overrange errors.

The *On-Chip Data RAM* is a fast memory that stores signal data and coefficients used for the immediate computational needs of the processor. Having this on-chip storage space speeds up the computational process by avoiding the need to access slower outside memory systems or data except when specifically needed for immediate processing.

**Program Bus Subsystems**

The *Program Bus* interfaces with the subsystems that control program execution and "housekeeping."

The *Program Memory* contains the sequence of instructions which constitutes the particular program running on the chip. Programs will vary with each application of the device for a different signal processing task. The program can be stored in the on-chip Read Only Memory (ROM) or another memory external to the chip.

The *Program-Counter and the Stack* control the execution of the program in the program memory by sequencing through the memory the addresses and keeping track of the branches, subroutine calls, and interrupts which occur during the course of the program.

*(continued on page 53)*

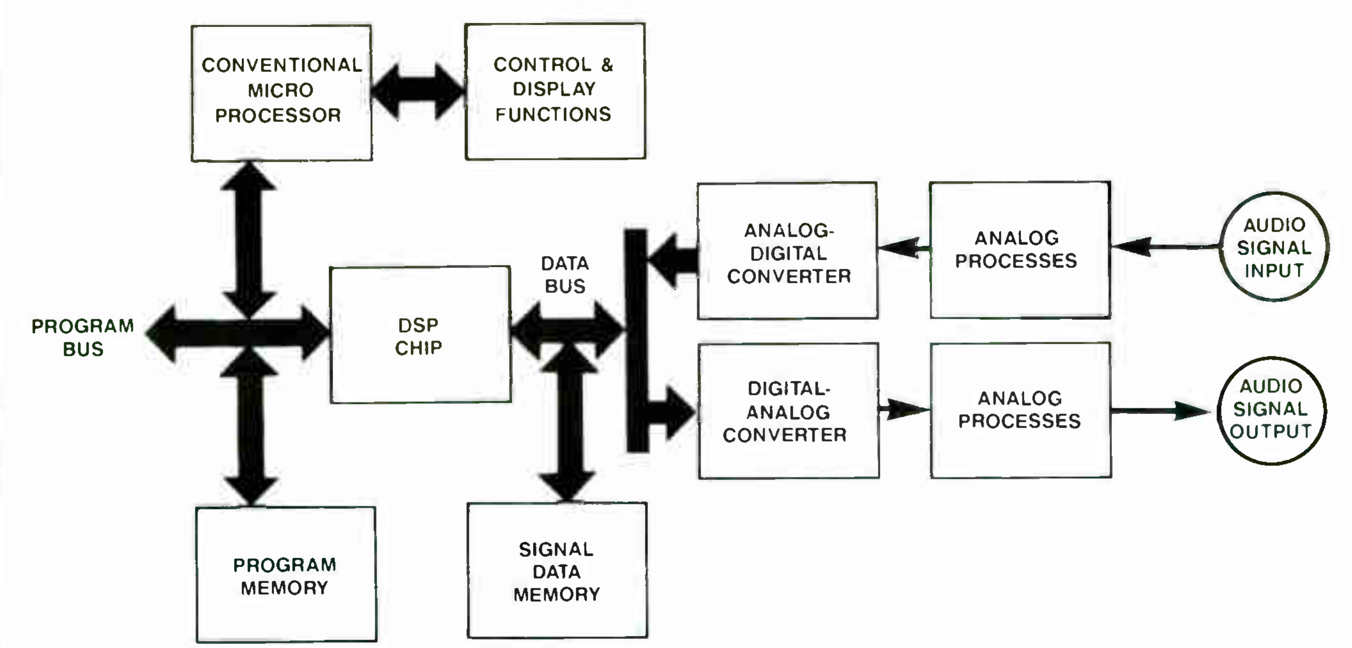
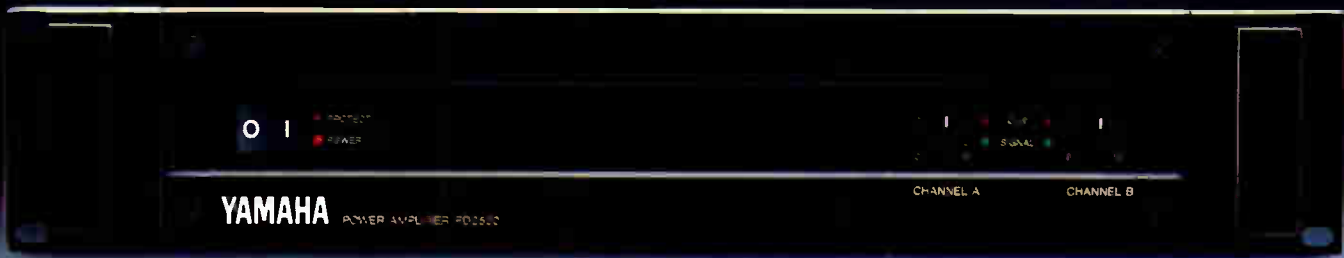



**Figure 2**: System Diagram for Generic DSP Audio Product

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Installation Profile

**By John Linde and Mel Smith**

# Marriott's Orlando World Center Resort And Convention Center Ballroom Combining System

The mission was to provide a more modern, more versatile and more reliable ballroom combining system to a company used to state-of-the-art equipment.

Imagine the problems inherent in designing and installing a ballroom combining system for more than 45 separate rooms. Now add to this the requirement to have a central ballroom combining system that taps into these individual systems, all over an area of 225,000 square feet. That is precisely the system that has just been installed at the Marriott's Orlando World Center Resort and Convention Center in Orlando, FL.

The Marriott facility has multi-purpose ballrooms and meeting rooms and the operators place heavy demands on their facilities, prompting their sound system designer and selected contractor to seek superior performance at a reasonable cost. In the past, sound system designs for hotel ballroom and meeting room systems were dictated by developers, architects and consultants trying to anticipate and interpret the needs of the operators. Direction and expectations offered by the hotel staff were typically based on incomplete knowledge and preconceived notions of the limitations of available technology. Adding to these distractions were an abundance of conflicting interests inherent in the design, procurement and installation.

### The Consultant/Designer

Marriott's Jeff Loether is design manager of audio-video engineering at Marriott Hotel's architecture and construction division in Washington, D.C. and acted as the consultant and designer of the system. He has done a wide variety of consulting and contracting work on such applications as background and live performance sound systems for places such as the Marriott Hotel chain and others.

His goal was to have a system that was highly intelligible and reliable, utilizing sound audio design techniques. It was important to him, as well as to others in the Marriott Hotel chain, to have standardization of equipment, terminations and labeling, as well as making maintenance and testing very easy.

As a result of this goal, the components that were selected for use in the system needed to display a great deal of versatility, architectural design and ease of repair.

His experience pointed to the use of equipment such as Altec Lansing and FSR because their architectural design matched the needs of this application quite well.

### System Overview

Marriott's Orlando World Center Resort and Convention Center site consists of a cluster of multi-use ballroom and meeting room sound systems. Three central control stations are located in the convention center building with all of the individual multi-purpose ballroom and meeting room sound systems in this complex hard-wired into the control consoles. The separate locations, have individual sound systems with remote override capabilities from the central control units. The use of the FSR ML-132 Ballroom Combining System signal-activated module eliminated the need for any control circuits to enable the override capability.

The design of the ML-132 system takes advantage of current technology to provide a blend of reliability, ease of use, flexibility, and reasonable cost. Computer control provides operation of music and microphone functions of up to 16 rooms. When any of the rooms are combined by the press of a button, their independent sound systems become one. Each control panel, speakers, amps, etc. will automatical-

**(Below) The system is wired using ADC Pro Patches. (Facing page, above) The Marriott Orlando World Center. (Facing page, below, left) Racks that make up the system. (Facing page, below, right) FSR ML-132 being used in a combining system logic control unit in a control panel that combines all the rooms.**

ly operate as one sound system. No physical joining of components is necessary. Any number of rooms can be combined and there can be any number of combined groups.

This system provides what past systems have failed to accomplish. The ML-132 eliminates the need for skilled operators and can be used by your own staff with minimal instruction. The entire system can be controlled from one location as well as allowing specific functions to be controlled from each room. The quality of your sound system is not reduced at all by this system as it has been in past systems. Sound engineering practices were adhered to throughout the system, resulting in excellent isolation and insuring minimal background noise. Due to the simplicity of operation, there is

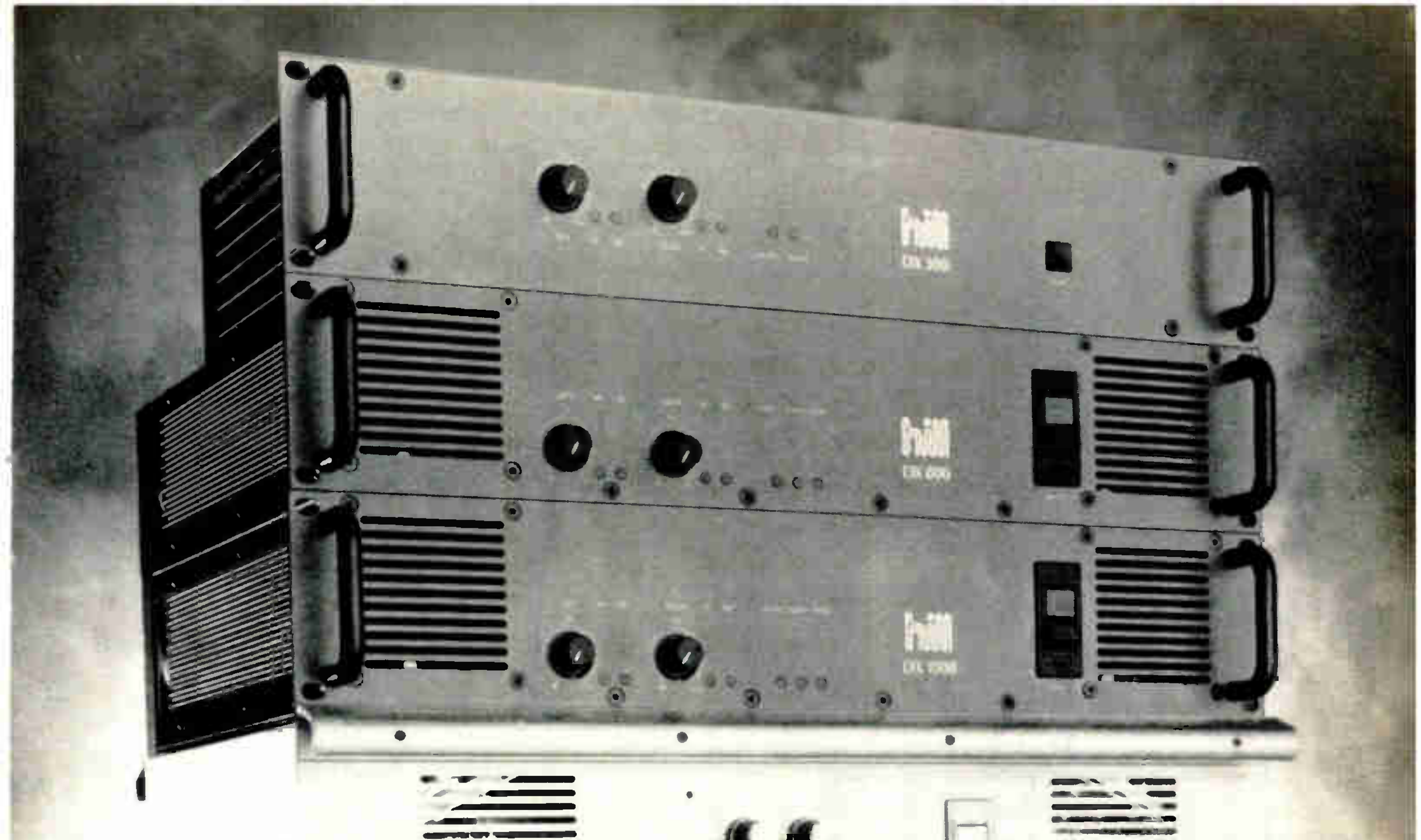

Photos by Bill Vargas

less chance of human error that would lead to diminished intelligibility. Built-in safeguards prevent system failure due to misuse or failure of one component.

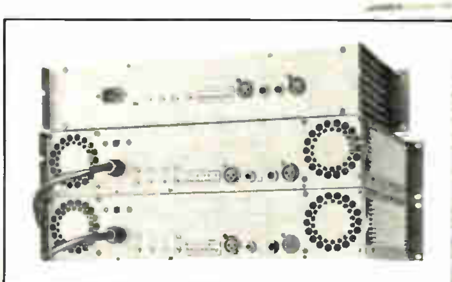
The entire system includes: a wallplate for each room; a ballroom control panel; a monitor/VU panel; a logic unit; and audio matrix used in conjunction with Altec automatic mixers; a music speech unit; a monitor/VU switcher; and a connector block. The system, including the ballroom and monitor panels, fits into one equipment rack approximately seven feet high by nineteen inches wide. The system is easy to install and all connections are clearly labeled. Where earlier systems took weeks to install, this entire system can be installed and placed in operation in one day.

In each room, a wallplate panel controls music (OFF, SOFT, MED., LOUD) and microphone functions. The microphone switch is only operable with a safety key switch to avoid accidental interruptions. All switches are marked and have lamp indication.

The Ballroom Control Panel is the master panel and can be mounted in the equipment rack. On this panel, there are switches to combine rooms, control microphones, and operate the head table speakers. Lamp indication shows when music is playing in a room. Any number of adjacent rooms can be combined by operating the MIX switches between the room areas on the panel. An INTERLOCK switch must be held to operate any MIX or LOCAL INPUT (or MICROPHONE) switches preventing accidental combining. The HEAD TABLE switch operates without the use of the INTERLOCK switch. With the HEAD TABLE switches, the speakers above the head table locations in a room can be shut off when a microphone is to be used at that table. This prevents feedback. If this switch is activated, the speakers will automatically turn off when the Local Inputs become active. When the music is turned on, the speakers will also come on. Any number of HEAD TABLE switches can be provided for a room. If the HEAD TABLE switch and the local input in that zone are both active, the HEAD TABLE switch is disabled, preventing an operator from inadvertently turning the speakers back on and inducing Feedback.



# introducing the DX300, DX800 and DX1500 power amplifiers



The DX300, DX800 and DX1500 amplifiers all feature balanced XLR inputs, balanced and unbalanced ¼" inputs, banana outputs, ground lift switch and a terminal block with access to all inputs and outputs.

**DX300** **output power:** **200** watts into 8Ω (per channel, both channels driven, 20Hz - 20kHz, -0.5dB) **400** watts into 8Ω, **450** watts into 4Ω (burst power*) **distortion** (250mW to rated power at 8Ω): IMD SMPTE: < 0.01%. THD (1kHz): < 0.01%. THD (20kHz DIN): < 0.02% **size:** 2 rack spaces, 8½" behind front panel **weight:** 7Kgs, 16 lbs. **cooling:** 2 passive heatsinks.

**DX800** **output power:** **250** watts into 8Ω, **400** watts into 4Ω (per channel, both channels driven, 20Hz - 20kHz, -0.5dB), **800** watts into 8Ω (bridged mono), **800** watts into 4Ω, **900** watts into 2Ω (burst power*) **distortion** (250mW to rated power at 8Ω): IMD SMPTE: < 0.01%. THD (1kHz): < 0.01%. THD (20Hz-20kHz DIN): < 0.02% **size:** 2 rack spaces, 13" behind front panel **weight:** 13Kgs, 29 lbs. **cooling:** 1 servo controlled DC fan.

**DX1500** **output power:** **300** watts into 8Ω, **500** watts into 4Ω, **750** watts into 2Ω (per channel, both channels driven, 20Hz - 20kHz, -0.5dB), **1000** watts into 8Ω, **1500** watts into 4Ω (bridged mono) **1500** watts into 2Ω, **1600** watts into 1Ω (burst power*) **distortion** (250mW to rated power at 8Ω): IMD SMPTE: < 0.01%. THD (1kHz): < 0.01%. THD (20Hz-20kHz DIN): < 0.02% **size:** 2 rack spaces, 13" behind front panel **weight:** 15Kgs, 34 lbs. **cooling:** 2 servo controlled DC fans.

*Burst power is a 1kHz tone for 10ms every 100ms, single channel (an indication of the amplifiers ability to handle music transients and tolerate deviations in nominal speaker impedance)

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**Hill Audio**

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The Monitor Panel, which also can be mounted in the equipment rack, allows supervision and control of the audio functions in each room. What is being heard in a room can be heard at the Monitor Panel through the speaker there. The VU meter will indicate the sound level and the status indicator shows exactly what switch has been selected for a room.

Power for the system comes on with the rack power. The Logic Unit, the Audio Matrix and the Music Speech Unit are plugged in at the rack.

In total, the installation used 162 Altec 920-8 12-inch coaxial full-range paging loudspeakers, 38 Altec 1674 and 10 Altec 1678B automatic microphone mixers, 60 Altec 2271 power amplifier modules and eight Altec 2280A incremental power amplification systems, and three FSR 132 series hotel combining systems. The main control consoles consist primarily of one FSR ML-132 hotel combining system with its monitor/VU-meter panel, Altec automatic microphone mixers, Altec power modules, Altec incremental power amplification systems, a UREI 1176 LN peak limiter, a UREI 535 and 5547/c graphic equalizers and ADC BJF 205 patchbays. The ML-12 was custom-built by FSR for this installation.

**Contractor Selection**

Once the systems were designed and agreed to, the process began to find a contractor to install it. Invitations to bid were sent to a selected group of qualified bidders. They were initially asked to attend a meeting where the systems were overviewed. These contractors then submitted preliminary bids that enabled the Marriott to narrow the group down to three companies. The bids were reviewed by the consultant and Marriott. As a final step, oral interviews were conducted by the consultant, Marriott facilities engineering and Marriott management with each of the three candidates.

The Danforth Company, Longwood, FL., was selected to install the systems.

Their initial concern, and the focus of their planning, was the logistics associated with an installation covering such a large area. It also presented interesting challenges in assuring that quality signals could be passed over lines without degradation. They spent a considerable amount of time testing and verifying the line quality to make certain that the completed systems were operated as designed, said Richard Megow, engineer for the Danforth Company.

**System Installation**

Work began with the installation and impedance testing of all speaker lines and installation and impedance testing of central control lines and their outputs. Frequency response tests for all signals generated were required and performed as well.

Once these tests were completed, assembly of the individual multi-use ballroom and meeting room combining systems was started. These systems were staged in their shop in Longwood, FL., where they were checked out and prepared for installation. Once the checkout was complete, the systems were transported to the Marriott site and installed.

As previously stated, a key component of the system was the ability of separate locations to receive program signals from the central controls via hardwired lines. To facilitate this communication link, special control unit/routing switches were developed to assure quality communications.

Four factors were considered in development of these systems: serviceability, dependability, performance

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and durability. Serviceability was addressed by having the products Danforth developed located at the central controls and Altec/FSR circuits for the separate locations mounted in modular edge-card frames at the central controls. This allows the cards to be interchanged, replaced or serviced without shutting down the entire system, a special consideration of importance in these multi-use ballroom and meeting room combining systems.

To achieve dependability and performance, the transformers were by Soundolier. They incorporated Soundolier HT-87 8-watt line transformers (70.7 volts) and Soundolier HT-167 high quality line transformers (16-watt).

All line inputs and outputs in the building were tagged. Danforth chose to fabricate the special control unit/routing switches (amplifier subsitution panels) themselves to make certain that all factors were met. They took the design and breadboarded the circuits for test. Once testing was complete, the circuits were manufactured, finished and installed. All boards had threaded pop studs and gold contacts were on all relay and patch points for good electrical integrity. All racks go to XIT grounds outdoors, that are designated for the sound systems, so as to suppress line transients, particularly important to operations in Florida because of the heavy thunderstorm activity.

**Nearing Completion**

The final phase of the installation was the assembly and testing of the central control units. All pieces such as the line driver cards, mixers, amplifiers and other components were assembled into 14 Sondolier electronic equipment cabinets while the FSR units were being finished. Once these components arrived, they were installed. An advanced wiring team arrived to tie the central control areas together. All lines were tested to determine line levels once the new systems were connected. Finally, the line levels were set and final testing completed on the central control areas. Following this, the entire system, location-by-location, was checked to assure that the system was performing as required.

Jeff Loether and the Marriott wanted to make certain that any future changes or additions to the system continue to follow the standards that have been set here. This will provide a level of continuity and uniformity as the Marriott's needs expand.

The design, customer involvement and careful installation work have all combined to produce a complex system that meets the customer's needs now, with room to grow in the future. Lessons can be learned from this close interaction of consultant, customer and contractor. When such cooperation is demonstrated and good communication maintained, all can focus their time and energy on doing their jobs well and not waste time over needless disagreements and misunderstandings.

As Megow said, the installation was not without its trying moments, but in general, it went smoothly. They have to credit the fine sense of cooperation shown by all involved for this to have happened.

---

**John Linde is a freelance writer for TV, radio, theater and has had work published in numerous national magazines. He's a screenwriter who's film, "Before God" is currently in production.**

**Mel Smith is president, senior consultant and video engineer for Audio Design Group Inc. of Orlando, FL. He holds a BS degree in Engineering.**


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# S&C's Job Report

**Format**
**STATE**
*city:* Name of Job, $ Total of Construction, Phase of Project. Contact: Name, Company, City, State; Telephone Number.

**TOTAL CONSTRUCTION**
1—up to $1 million
2—$1 million to $9 million
3—$9 million to $17 million
4—$17 million to $25 million
5—$25 million and up
NA—Not Available

**PHASE OF PROJECT**
A—Planning = Consultant is designing system
B—Pre-Bid = Final plans near completion
C—Bidding = Bid date set
D—Starting = Electrical Contractor/ General Contractor/ Owner buying now

**The following jobs are in various phases leading up to bid. If you are interested in any of the projects, please contact only the names printed below.**

### CALIFORNIA
***Bakersfield:*** Bakersfield Christian Life Center, 3, D. Contact: Neil A. Shaw, Paul S. Veneklasen & Associates, Inc. Santa Monica, CA; (213) 450-1733.
***Beverly Hills:*** Ma Maison Hotel, 4,C. Contact: Neil A. Shaw, Paul S. Veneklasen & Associates, Inc. Santa Monica, CA; (213) 450-1733.
***Cerritos:*** Performing Arts Center, 4,A. Contact: Robert Long, Theatre Projects, New York, NY; (212) 873-7211.
***El Segunda:*** Aerospace Building A-8, 5,C. Contact Neil A. Shaw, Paul S. Veneklasen & Associates, Inc., Santa Monica, CA; (213) 450-1733.
***Escondido:*** Escondido City Hall, 3,C. Contact: Neil A. Shaw, Paul S. Veneklasen & Associates, Inc., Santa Monica, CA; (213) 450-1733.
***Los Angeles:*** Simon Wisenthal Center, 3, A. Contact: Neil A. Shaw, Paul S. Veneklasen & Associates, Inc., Santa Monica, CA; (213) 450-1733.
***Ojai:*** Ojai Valley Inn, 5,A. Contact: Neil A. Shaw, Paul S. Veneklasen & Associates Inc., Santa Monica, CA; (213) 450-1733.
***Pasadena:*** Lake Avenue Congregational Church, 4,A. Contact: Neil A. Shaw, Paul S. Veneklasen & Associates, Inc., Santa Monica, CA; (213) 450-1733.
***Sacramento:*** Mercy Hospital, 2,B. Contact: Neil A. Shaw, Paul S. Veneklasen & Associates, Inc. Santa Monica, CA; (213) 450-1733.
***San Francisco:*** St. Mary's Cathedral, NA,A. Contact: Marc Beningson, Jaffe Acoustics, Norwalk, CT. (203) 838-4167.
***Santa Monica:*** Santa Monica Bay Hotel, 5,A. Contact Neil A. Shaw, Paul S. Veneklasen & Associates, Inc., Santa Monica, CA (213) 450-1733.
***San Jose:*** Fairmont Hotel, 5,D. Contact: Neil A. Shaw, Paul S. Veneklasen & Associates, Inc., Santa Monica, CA; (213) 450-1733.

### CONNECTICUT
***Hartford:*** Connecticut State Capitol Hall of the House of Representatives, NA, C. Contact: Mark Beningson, Jaffe Acoustics Inc., Norwalk, CT; (203) 838-4167.

### FLORIDA
***Naples:*** Naples Performing Arts Center, 4,A. Contact: Robert A. Lorelli, Brannigan-Lorelli Associates, Inc., New York, NY; (212) 420-8787.
***St Petersburg:*** Bayfront Center Auditorium Renovations, 3,B. Contact: Robert Long, Theatre Projects, New York, NY; (212) 873-7211.

### ILLINOIS
***Highland Park:*** Ravinia Young Artists Insititute, 2,C. Contact: Chuck McGregor, Jaffe Acoustics, Inc., Norwalk, CT; (203) 838-4167.

### KENTUCKY
***Alexandria:*** Campbell County H.S. Gymnasium, 1,B. Contact: Richard J. Lemker & Associates, Covington, KY; (606) 261-9529.
***Covington:*** Holmes High School Auditorium, 1,C. Contact: Richard J. Lemker, Lemker & Associates, Covington, KY; (606) 261-9529.

### MISSOURI
***Mokane, Callaway County:*** South Callaway R-2 School District, NA, C. Contact: J. T. Weissenburgger, Engineering Dynamics International, St. Louis, MO; (314) 991-1800.

### NEBRASKA
***Lincoln:*** Lied Center for the Performing Arts, 4,D. Neil A. Shaw, Paul S. Veneklasen & Associates, Inc., Santa Monica, CA; (213) 450-1733.

### NEW YORK
***Astoria:*** American Museum of Moving Images, NA,B. Contact: Marc Beningson, Jaffe Acoustics, Norwalk, CT; (203) 838-4167.
***Jamestown:*** Palace Theater, 2,B. Contact: Robert A. Lorelli, Brannigan-Lorelli Associates, Inc., New York, NY; (212) 421-8787.
***New York:*** John Jay College for Criminal Justice, 5,C. Contact:Robert Benson, Knudson-Benson Associates Inc., Mercer Island, WA; (206) 232-2273.
***New York:*** JP Morgan Bank Trust Committee Room, NA,B. Contact Marc Beningson, Jaffe Acoustics, Inc. Norwalk, CT (203) 838-4167.
***New York:*** Metropolitan Opera, NY Philharmonic Summer Parks Concerts, NA,A. Contact: Chuck McGregor, Jaffe Acoustics, Inc., Norwalk, CT; (203) 838-4167.

### OKLAHOMA
***Oklahoma City:*** Remington Park, 5,A. Contact: Neil Johnson, Ewing Cole Cherry Parsky, Philadelphia, PA; (215) 923-2636.

### OHIO
***Cleveland:*** Cleveland State Music Building, NA,A. Contact: Marc Beningson, Jaffe Acoustics, Norwalk, CT; (203) 838-4167.
***Cleveland:*** Palace Theatre-Playhouse Square, 2,C. Contact: Marc Beningson, Jaffe Acoustics, Inc., Norwalk, CT; (203) 838-4167.
***Columbus:*** Ohio State Office Technology Tower (Office) NA, C. Contact: Marc Beningson, Jaffe Acoustics, Inc., Norwalk, CT; (203) 838-4167.
***Columbus:*** Ohio State Office Tower (Theaters), 5,C. Contact: Chuck McGregor, Jaffe Acoustics Inc., Norwalk CT; (203) 838-4167.
***Columbus:*** Ohio State University Wexner Center for the Visual Arts, 5,D. Contact: Chuck McGregor, Jaffe Acoustics, Inc., Norwalk, CT; (203) 838-4167.
***Sharonville:*** Sharonville Municipal Building, 2,C. Contact: Richard Lemker, Lemker & Associates, Covington, KY. (606) 261-9529.

### SOUTH CAROLINA
***Columbia:*** University of South Carolina, Kogor Center for the Arts, 3,B. Contact: Chuck McGregor, Jaffe Acoustics, Inc. Norwalk, CT; (203) 838-4167.

### WASHINGTON, D.C.
***Washington, DC:*** National Council of Catholic Bishops Conference Center, 2,C. Contact: Marc Beningson, Jaffe Acoustics, Inc. Norwalk, CT; (203) 838-4167.

### NEW BRUNSWICK, CANADA
***St. John:*** Bicapital Theater Project, 2,A. Contact: Robert A. Lorelli, Brannigan-Lorelli Associates Inc., New York, NY; (212) 420-8787.

### NOVA SCOTIA
***Halifax:*** Art Gallery of Nova Scotia, NA,B. Contact Peter Terroux, Halifax, N.S. (902) 429-4616.
***Halifax:*** A/V system for City Council Chamber of Halifax, NA,B. Contact: Peter Terroux, Halifax, N.S. (902) 429-4616.

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# NSCA Contractor's

## EVENTS

**APRIL 5 (Sunday)**

**5:30 PM.** *Contractor Caper.* Kick-off reception for Contractor's Expo '87. Sponsored by independent sales reps.

**APRIL 6 (Monday)**

**8:00 AM.** *KEYNOTE.* Larry Estrin offers a multimedia look at Liberty's Weekend's electronics.
**10:00 AM.** *Computer Aided Design.* Ron Baker, Joiner Rose Group.
**10:00 AM-12:00 PM.** *CCTV Hardware Overview.*
**10:00 AM.** *How to Prepare a Winning Bid.* SPONSORED BY SOUND & COMMUNICATIONS MAGAZINE.
**11:00 AM.** *To Hum or Not to Hum.* Richard C. Cabot and Cal Perkins.
**11:00 AM.** *Business Planning and Operations.* Jess Santaularia.
**1:00 PM - 5:00 PM.** EXHIBIT FLOOR OPEN.

**APRIL 7 (Tuesday)**

**8:00 AM.** *Articulation and Intelligibility.* Ken Jacob, Bose Corporation.
**8:00 AM-12:00 PM.** *A Hi-Tech CCTV Overview.*
**8:00 AM.** *Job Survey Technology.* Vic Hall.
**9:00 AM.** *Making More, and then the Most of Your Market.* SPONSORED BY SOUND & COMMUNICATIONS MAGAZINE.
**10:00 AM.** *From the Drawing Board to the Catwalk.* SPONSORED BY SOUND & COMMUNICATIONS MAGAZINE.
**10:00 AM.** *Making Effective Advertising and Promotion Work for You.* SPONSORED BY SOUND & COMMUNICATIONS MAGAZINE.
**11:00 AM.** *The How and Why of Speaker Design.* Ted Uzzel, Altec.
**11:00 AM.** *A Practical Guide to Effective Use of Spreadsheets.* Dave Marsh.
**7:00 PM-9:00 PM.** *Faculty Forum.* Informal evening session with instructors.

**APRIL 8 (Wednesday)**

**9:00-12:00 PM.** EXHIBIT FLOOR OPEN.

**TOUR.** "Special contractor's and significant-other tour of the Superdome."

Once again the community of sound and communication professionals meets to exchange information and show product and technology at the NSCA convention. This year the annual convention of the National Sound and Communications Association is titled Contractor's Expo '87 and sets up in New Orleans from April 5 through April 8 at the Fairmont Hotel.

The seventh convention of its kind, this one also promises to be the biggest, with over 214 exhibitors and over 3,000 expected attendees in 40,000 square feet of space. (Last year's meet hosted, by comparison, only 173 exhibitors.)

The NSCA's amenities at this year's show include an opportunity for consultants, manufacturers, and reps to avail themselves of free meeting space in which to conduct sales meetings, hold training sessions, special events, pre-bid or bid conferences.

The availability of the meeting facilities indicates the serious business considerations at this show as the convention offers opportunities for serious business networking. The schedule of events reinforces that. *Sound & Communications* magazine is sponsoring four panel discussions within that schedule of events which will cover both technical and marketing concerns.

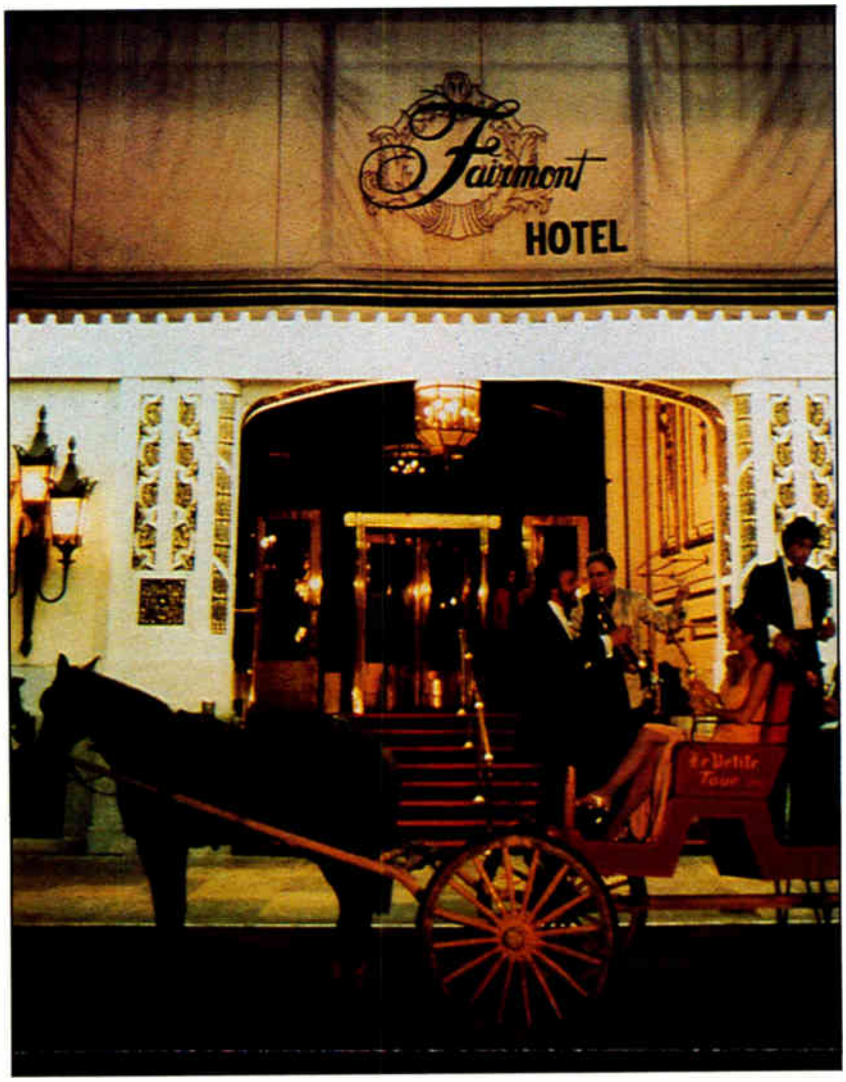
Once again, NSCA-TV News will be on the air, produced by the publishers of *Sound & Communications* magazine

# Expo '87

and written and edited by the editorial staff of *Sound & Communications* magazine. The television show is similar to others produced by Testa Communications at other, even larger conventions. It offers, on large-screen televisions on the floor of the show and in all the convention hotel rooms, a professional news show concerning only the Contractor's Expo '87, narrowing the broadcast band for concerns pertinent and serious for all those attending the Expo.

A complete list of the schedule of events of Expo '87 accurate at press time, follows. To highlight the events: On April 3 and 4 (prior to the show), a Salesman's Orientation Program will take place, a basic introductory course on sound for sales people. On April 6, 7 and 8, the NSCA Technical Track Workshops will be held. NSCA Special Workshops will be offered on April 6, 7 and 8, which will include advice on investment, marketing, negotiations, etc. Apri 6, 7 and 8 are also the dates of the NSCA Video Security Track Workshops for contractors currently installing or considering the video security market.

Be sure to register for the programs of your choice. Have a good show! In subsequent issues, *Sound and Communications* will give a thorough rundown of the product, technology and marketing innovations exhibited at this year's NSCA Contractor's Expo.



The Fairmont Hotel where this year's expo is being held.

## SPEECH SYNTHESIS

*(continued from page 28)*

the appropriate electronics.

### The Mechanics of Human Speech

The process of producing speech sounds is an extremely complex set of actions. Judson and Weaver in their book "Basic Speech and Voice" (College Typing Co., Madison, 1933), describe 30 single muscles, pairs, and groups of muscles, amounting to more than 100 in all, in their discussion of inspiration and expiration. The process of producing speech can be divided into four main categories: breathing; phonation; resonance; and articulation. In viewing these actions separately, we should be careful not to forget that they interact as a coordinated system.

The human speech mechanism is analogous to a wind instrument. The breathing organs provide the power for all the sounds of our language by producing an air flow from the lungs. In phonation (the production of vocal sounds), the vocal cords are set into a controlled vibration which then modulates the air stream of expelled air. These masses of air that are passed through the vocal tract (the throat, mouth, and nasal passages), are excited as a result of the vibrations of the vocal cords. The vibration of air in these enclosed volumes is often called resonance. This resonance can also be viewed as a set of filters. These filters are called *formants* and there are usually about three or so which can be moved up and down in frequency to reinforce various pitches. These segments of speech are called the *voiced sounds.*

The valve-like action of the tongue, lips, teeth, palate, and other structures in interrupting and modifying the vocal sounds and/or air flow is called *articulation*. Customarily, it refers principally to the production of consonants in which the movements of the articulators are more definite. Speech sounds are thus generated in three ways: voiced sounds are produced by the modulated air stream; fricative sounds are generated by forming a constriction at some point in the vocal tract, usually in the mouth, and forcing air through the constriction; plosive sounds result from making a complete closure, building up pressure behind the closure, and abruptly releasing it.

Vowels alone comprise just a small number of the sounds used in speech. However, vowel sounds do carry most of the energy of speech. Generally speaking, the vowels have higher intensities (usually from 12 to 20 dB higher) than the consonants. The vowels have spectral components that are lower in frequency than those of the consonants. Furthermore, vowels achieve a steady state for a longer time than consonants; in other words, consonants are more transient.

While vowels may contribute to most of the energy content in speech, the consonants contribute most to the identification of words and phrases. The consonants are further categorized by fricatives, plosives, nasals, glides, and semivowels. All of the distinct sounds that are used to produce vowels and consonants are called phonemes, of which there are some 40 in total in the English language. Phonemes can be identified by their spectrum. However, phonemes do not occur as individual events in time, but are rather slurred together, forming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what are known as dipthongs with a corresponding dynamic spectrum.

The rapidity of speech precludes the precise production of speech sounds. The ability of our system of speech communication to tolerate slight changes in articulatory placement for both vowels and consonants without affecting their value is therefore vital to speech comprehension. Since speech is not made up of sounds used individually, but of sounds put together in a series, this tolerance permits us to maintain values at high rates of transmission. In connected speech, the movement from one articulatory position to another is often as important as the position itself. This movement is called the transition. The spectrum of connected speech shows that much of speech is transitional.

Researchers have used speech spectrography and X-ray motion pictures to show that the articulations of adjacent phonemes interact and that transient movements of the vocal tract for the production of any phoneme lasts much longer than the average duration of the phoneme, *i.e.*, the articulatory gestures overlap and are superimposed. This transient nature of the vocal tract has perceptual ramifications as well. Much information about the identity of a consonant is carried not by the spectral shape at the "steady state" time of the consonant, but by its dynamic interactions with adjacent phonemes.

### Speech Synthesis

With the progress made in the field of speech synthesis through the 1970s, the technology has been established for the production of phonemes. Various schemes have been devised to produce this synthetic component of human speech; However, there are two obstacles: how to deal with the dynamic aspects of running, connected speech, and the large amounts of information needed to represent useful lengths of speech.

At least three techniques have evolved as the main tools for speech synthesis, namely: formant (resonant frequency) synthesis; linear predictive coding; and waveform sampling. Only the first two are used in text-to-speech conversion because they require less storage and slower data rates.

Formant synthesis, for example, requires a data rate of about 100 bits, based on a typical rate of 12 phonemes per second, with each phoneme characterized by an 8-bit code. By contrast, waveform sampling uses common A/D conversion and requires about 64,000 bits per second for uncompressed speech (8,000 samples per second to capture up to 4,000 Hz, multiplied by 8 bits per sample).

In formant synthesis, additional hardware digitally simulates resonance and antiresonance phenomena in the vocal tract. Arranged both in series and in parallel, circuits replicate the function of the vocal tract cavities. A periodic impulse emulates the modulated air stream, and a noise source emulates the fricative sounds. The impulses, as well as the circuit's resonant frequencies, bandwidths, and gains, are varied to simulate the movement of the vocal tract, tongue, lips, and teeth.

Introduced more than 15 years ago, linear predictive coding (LPC), by

*(continued on page 62)*

# BOOK REVIEW

*by Jesse Klapholz*

# The Great Beige Book

Davis, Don and Davis, Carolyn, *Sound System Engineering: Second Edition*, Indianapolis: Howard W. Sams & Co., 1987, $39.95 (hardcover).

Everybody in the sound business has been talking about the arrival of this book. Why the topic of conversation? What is so special about a second edition?

To place the state-of-the-art in perspective with the introduction of the first edition; ten years ago, when the first edition was released, bi-amping was a secret to most everyone. In fact, a common comment would have been, "Why that's when you hook up a second amplifier for a backup." Indeed, we have come a long way in learning about the technical aspects of sound reinforcement, and in large part through the efforts of the Davises.

The first edition of *Sound System Engineering*, often referred to, over the years, as the Great Yellow Book, was based on the text used in the early days of Syn-Aud-Con (a three-day basic audio seminar conducted by the Davises and others). The second edition contains new chapters and an entirely new look. Besides the color change (this one is beige and looks more like a textbook), the layout makes for easier reading, and there are comprehensive, valuable bibliographies at the end of each chapter. The readings referred to in the bibliographies are excellent technical backups, and in many cases offer good perspective on the respective technological evolution. While many of the texts are out of print, a trip to the local main library or university should turn them up.

The Table of Contents is also greatly improved, since it is page-keyed in detail, topic-by-topic. When used as a reference, this will save time in finding information. A total of light chapters has been added; some are expansions of chapters found in the first edition; and four articles have been added in the appendix. On the other hand, a few features have been dropped. The appendix containing a comprehensive listing of basic equations used in audio work has been omitted, as well as the test questions and answers sections.

The original chapter on the decibel notation system has been greatly expanded into two chapters: "Mathematics for Audio Systems," and "Using the Decibel." By starting with a tutorial review of mathematic basics,

## *"The reviewer is sure that the minimal investment is worthwhile for those who need a quick reference to almost any aspect of sound system design."*

the reader has at his disposal an adequate reference to understand what follows. The chapter on the decibel, while it contains basic information that has not changed, benefits from the new layout look and extra graphics. The same information is much easier to read, understand, and present to others.

The first edition's chapter on system interfacing has been broken up into three chapters: chapter 5, "Interfacing Electrical and Acoustic Systems;" chapter 12, Microphones;" chapter 13, "Loudspeakers and Loudspeaker Arrays." The theme of chapter 5 is gains and losses, and covers gain structure and total power output. Chapter 6, "Loudspeaker Directivity and Coverage," has been expanded to include a short discussion on acoustic centers and apparent apex, an outlined discussion on the various competing methods of measuring loudspeaker Q, and a detailed section on the architectural mapping methods of loudspeaker/loudspeaker arrays.

The following three chapters, "The Acoustic Environment," "Large-Room Acoustics," and "Small-Room Acoustics," all grew out of a single chapter in the first edition. The new chapters cover more practical ground and basics in acoustics and are useful to the contractor in understanding how his sound system will interface with its environment. Similarly, the original chapter on "Designing for Acoustic Gain" has been expanded and is now preceded by a chapter entitled "Designing for Speech Intelligibility."

The remaining chapters deal with the final design, installation, and testing of the system. Again, through more complete and up-to-date information, these basic techniques have been presented in a more useful form. While at first glance the many TEF plots may seem profusely technical, the intention is to explain the basics of audio/acoustic diagnostics.

In the over-15 years since we have seen the beginnings of *Sound System Engineering*, there has certainly been a lot of growth and maturity in the sound and communications business. This growth is reflected in the almost doubling in size of the Davises' book. This reviewer is sure that the minimal investment is worthwhile for those who need a quick reference to almost any aspect of sound system design.

## DSP CHIP

*(continued from page 34)*

The *Controller* handles timing, interrupt, and other functions that are necessary for the device to operate as a fully functioning microcomputer.

To understand the other system subsections in the TMS32010 device, or to gain a more comprehensive insight into every aspect of this particular chip, refer to the *TMS32010 User's Guide*, available from Texas Instruments.

### Programming and Systems

#### Software development

The T.I. chip is only one of a number of different products of varying levels of performance and complexity. Needless to say, the process of designing products with complex systems like these requires a level of technical support greater than with previous electronic systems of almost any type.

Since the DSP chip is a computer, it needs a program in order to perform any function. A program for a DSP chip, at least at this point in time, is usually written in Assembly Language instructions appropriate to the specific processor used.

High-level languages like Fortran, Pascal or "C" are not as yet used frequently for two reasons. First, the necessity for very fast program execution and very compact programs dictates the use of assembly language coding; second, high level language support systems for DSP chips are not commonly available, though we can expect them soon.

The predominance of assembly language programming makes the art of design with DSP chips more demanding and arcane. Many types of functions implemented by digital signal processing could be implemented in relatively standard forms in high-level languages. The development of additional software support systems for DSP in general and for each chip in particular will ease the design process as time passes.

#### Peripheral Components

The DSP chip by itself is not sufficient to perform all the functions involved in a complete digital audio product. Figure 2 depicts a simplified block diagram of a digital audio system using a DSP chip.

*(continued on page 74)*

by William R. Thornton, Ph.D., P.E.
Thornton Acoustics & Noise

# A Review of Sound Energy

In several previous issues of *Sound & Communications* we have talked about sound intensity, power, and the *master equation*. In this issue, we will review the key ideas from the previous columns.

### Work and Power

You will recall that *intensity* and *sound pressure* are interrelated. Work is defined as force times distance. In acoustics, pressure times area is force, and work is expressed as:

(1) Work = [P(t)A]dr
where:
P(t) = pressure
A = area
dr = distance

Power is defined as the rate of work per unit of time:

(2) Power = d(Work)/dt = [P(t)A]dr/dt

Intensity is defined as P(t)V(t), and equation

(3) Power = [P(t)A]V(t)

can be rewritten as:

(4) Power = P(t)V(t)A = I(t)A
where:
I(t) = intensity

### Pressure, Intensity, and Power

Pressure, Intensity, and Power are key components in acoustics. "Sound pressure level" is the time-averaged pressure magnitude of a sound wave expressed in a decibel ratio to a reference. Its magnitude varies with distance. It is a *scalar* quantity; *i.e.*, it has magnitude but no direction.

"Sound intensity level" is the time averaged power per unit area emitted by a sound source. The magnitude of intensity is expressed in terms of decibels and it is a vector quantity; *i.e.*, it has magnitude and a direction. It measures the flow of power per unit area in a specific direction.

"Sound power level" is the integral (summation) of intensity times area for the entire surface area surrounding a sound source. It is the total energy radiated from the sound source per unit of time. It does not vary with distance; it is a fixed quantity for the source. Power is:

(5) $W = I_1A_1+I_2A_2+I_3A_3...+I_nA_n$
where:
W = sound power in watts
$I_n$ = intensity for the nth surface area
$A_n$ = nth surface area

Power is computed from the normal component of intensity, *i.e.*, normal to the surface area.

### Sound Intensity Method

In the past, intensity and power were used primarily in the laboratories and classrooms rather than in the field. With advances in signal processing and FFT instrumentation, an old idea was reborn. Two microphones are used with a modern FFT system to measure sound intensity. It is measured using the "cross spectral density function." The imaginary part of this function is directly proportional to net sound intensity.
Sound intensity is:

(6) $I(f) = P_1P_2\sin(\varnothing_{12})/(4dr\pi f)$
where:
I(f) = intensity at frequency f
$P_1$ = mean square pressure of mic 1
$P_2$ = mean square pressure of mic 2
dr = distance between mics 1 and 2
$\varnothing_{12}$ = phase angle of the acoustic signal
f = frequency

Sound power is computed by integrating intensity times area. Subject to reasonable limitations, the power output and directivity of a source are determined with confidence.

### Master Equation

Finally, these quantities are unified with the Master Equation for design work and analysis in the field. The Master Equation is:

(7) $Lp = Lw+10\text{Log}[(Q/4\pi r^2)+(4/Sa)] + 10.3$
where:
Lp = sound pressure level
Lw = sound power level
Q = directivity index
S = room area in feet
*a* = absorption coefficients

Sound pressure level is frequency dependent because directivity and absorption are frequency dependent. It is normally computed for narrow, one-third octave, or 1 octave bands.

### Direct and Reverberant Field

The Master Equation contains two important terms. The first term (within the log expression) describes the direct field effect. The numerator has directivity information which accounts for the radiation characteristics of the source. The denominator has distance information which describes the spherical spreading of the wave front. It is the area of a sphere. As distance increases, sound pressure decreases because the area increases. Therefore, pressure decreases inversely with area. The second term accounts for room effects. The room causes a sound buildup that is a function of room-size and absorption. At large distances, *i.e.*, well beyond the critical distance, the direct field diminishes and the reverberant field dominates. Sound bounces around randomly throughout the room and the wave fronts add together to produce a uniform sound field at all points in the room at large distances.

### Summary

This article ties together the important concepts that should be used in the design of sound systems. Quantitative tools such as the Sound Intensity Method are used to properly diagnose and solve problems instead of relying on rough estimates.

**Bill Thorton, president of Thorton Acoustics and Noise, holds a B.S.M.E. and an M.B.A. from the University of Pittsburgh.**

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# LAB TEST REPORT

# REVIEW OF BES C-70S

by Farrel M. Becker
Audio-Artistry

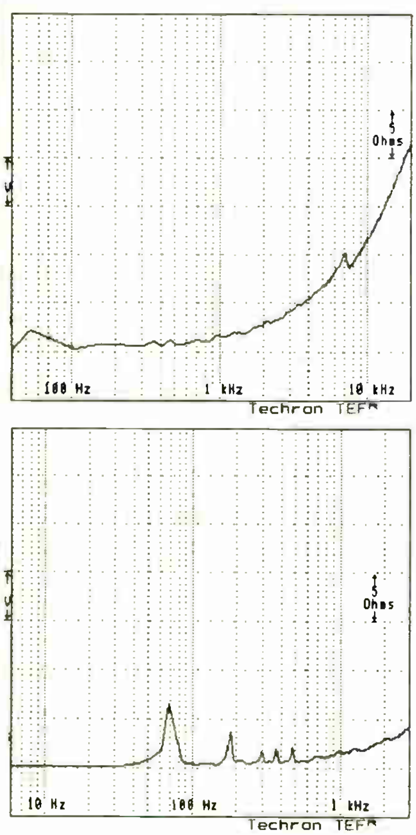



*Figure 1. Magnitude of impedance of BES C-70S, 0-20 kHz, 7 Hz resolution.*

Anyone who is in the professional audio industry and has not heard about the loudspeakers manufactured by Bertagni Electroacoustic Systems (BES) must have just returned from an extended trip to the moon. These speakers are being used in everything from distributed ceiling systems to large central cluster arrays. The August 1986 issue of *Sound & Communications* ("Contractor Closeup") featured a photograph of a huge spiral-shaped central array consisting of *204* BES loudspeakers.

Although BES manufactures a high fidelity line of loudspeakers, the company's professional line is primarily intended (or at least was *originally* intended) for use in distributed ceiling systems. The design of these speakers is unique. They are constructed primarily of what appears to be styrofoam with an aluminum frame. The aluminum provides a rigid support for the magnet assembly and the foam "diaphragm." They are definitely not anything like cone loudspeakers. They have flat fronts and come in several

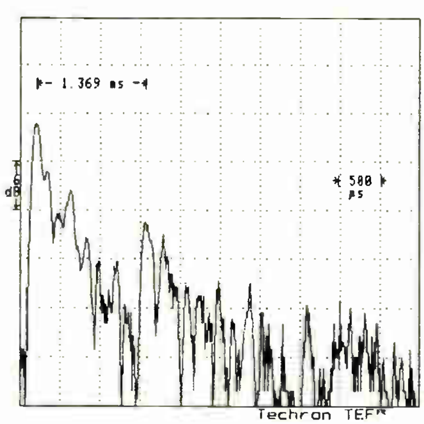



*Figure 2. On axis ETC.*

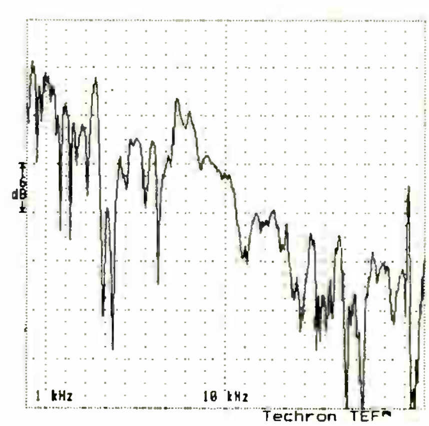
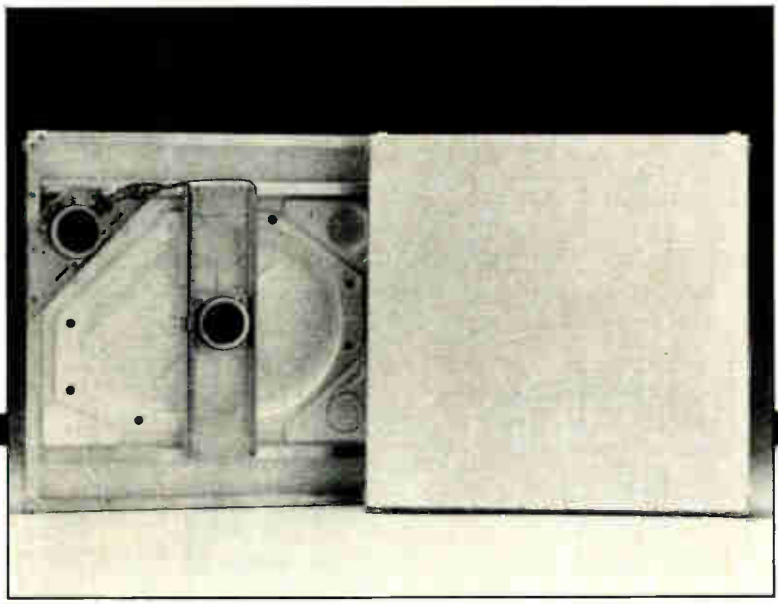



*Figure 3. On axis frequency response, 0-20 kHz, 150 Hz resolution.*



sizes. The C-70S is a full range, single driver loudspeaker. A two-way model, designated C-70D, is also available.

The C-70S is 23-and-five-eighths inches square and three inches deep. It weighs seven-and-a-half pounds and is designed to drop directly into a standard suspended ceiling T-bar grid system. BES provides several surface finish options which make the face of the loudspeaker look just like the ceiling tiles around it. This allows for very easy and rapid installation.

A little caution is probably called for when using this type of loudspeaker. If no back box is used, any debris, wiring, or other material that might be laid on top of the loudspeaker (either accidentally or out of ignorance by other trades) could result in objectionable buzzing and rattling. Since these loudspeakers can be installed quickly, it might be a good idea to wait and install them after the other trades have completed their work in the ceiling.

The C-70S carries an EIA power handling rating of 60 watts "RMS" and 100 watts for program material. Connections to the loudspeaker are made via two color-coded wires approximately nine-and-a-half inches in length. The ends of these leads are stripped and tinned. A positive voltage applied to the red lead (with respect to the black lead) produced a positive acoustic pressure in front of the loudspeaker.

The magnitude of the loudspeaker's impedance is shown in Figure 1. The impedance at 20 Hz is 5.2 ohms. It rises to 6.8 ohms at 1 kHz and 26.1

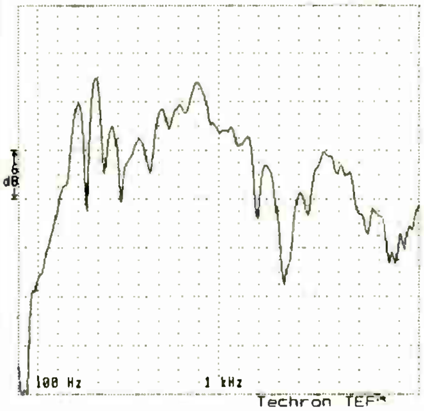



*Figure 4.* ***On axis frequency response, half-space, 0-2 kHz, 45 Hz resolution.***

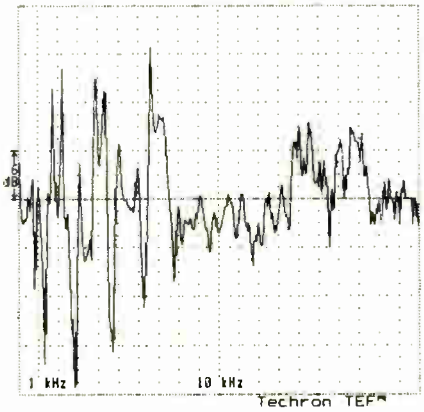



*Figure 5.* ***Difference between on-axis and 30 degrees off axis frequency responses, 0-20 kHz, 150 Hz resolution.***

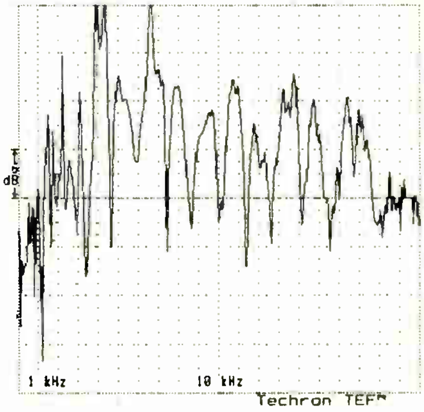



*Figure 6.* ***Difference between on-axis and 60 degrees off axis frequency responses, 0-20 kHz, 150 Hz resolution.***

ohms at 20 kHz. Notice the five resonant peaks at 70, 180, 293, 368 and 473 Hz. An additional peak occurs at 7,217 Hz. These peaks may indicate the resonant frequencies of the various areas of the panel. The back side of the foam diaphragm is molded into what appears to be separate radiating areas.

Figure 2 is an Energy Time Curve (ETC) of the C-70S. The time response of this loudspeaker fills the entire five milliseconds of the display. Frequencies up to 4 kHz are included in this long decay time. What appears to be a second arrival at 1.3 ms after the initial arrival is all part of this one loudspeaker.

The on-axis anechoic frequency response is shown in figure 3. The 4 foot/1 watt sensitivity at 1 kHz is 87.9 dB. The EIA sensitivity rating is 40.4 dB. Figure 4 shows the response at low frequencies with the loudspeaker in a half-space mounting (as would be the case when mounted in a ceiling). The response is usable down to 200 Hz. A frequency response, as that of the C-70S, results in a phase response that is difficult to interpret and is therefore not shown.

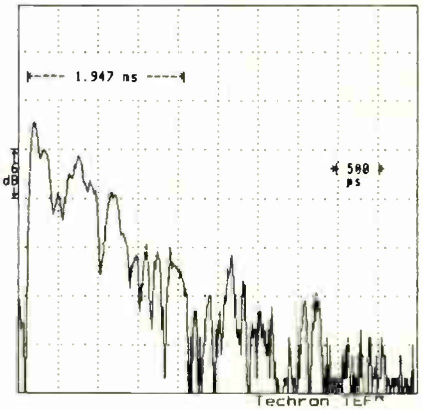



*Figure 7. ETC 30 degrees off axis.*

Figures 5 and 6 show the *difference* between the on axis response and the response 30 degrees and 60 degrees off axis respectively. In these figures, the dark line across the center of the display represents the normalized frequency response on axis. The figures show how the level increases or decreases for each frequency with reference to the level on axis. It can be seen that the response off axis is quite variable. Figure 7 shows an ETC taken 30 degrees off axis. The time response still fills the entire display. The loudspeaker takes almost 2 ms to emit the majority of the energy.

Figure 8 is the polar response for the

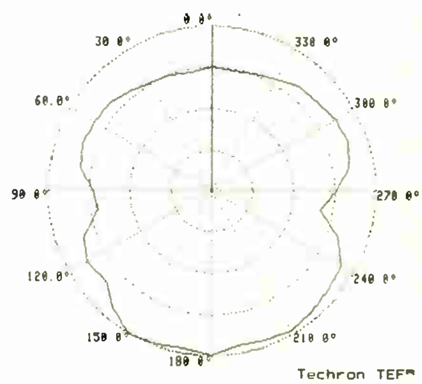



*Figure 8. 2 kHz octave band polar response, 10 degrees/data point, 10 dB/division.*

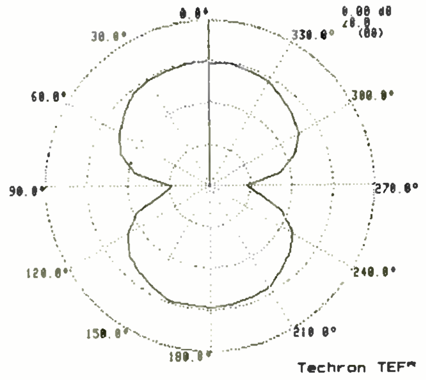



*Figure 9.* ***500 kHz octave band polar response, 10 degrees/data point, 10 dB/division.***

2 kHz octave band. The pattern is roughly bi-directional in shape. Notice that the sensitivity is almost 10 dB higher 180 degrees off axis than it is on axis. The on-axis Q is 0.22. The Q is less than one because of the increased sensitivity off axis. If the loudspeaker were mounted such that none of the energy from its back side entered the room, the Q would be 0.37. However, if the loudspeaker were mounted upside down with the front aimed up into the ceiling plenum and the back side into the room, the Q would be 4.42!

Figures 9 and 10 show the polar response for the 500 Hz and 8 kHz octave bands respectively. At lower frequencies the response is bi-directional. At higher frequencies the response approaches omni-directionality.

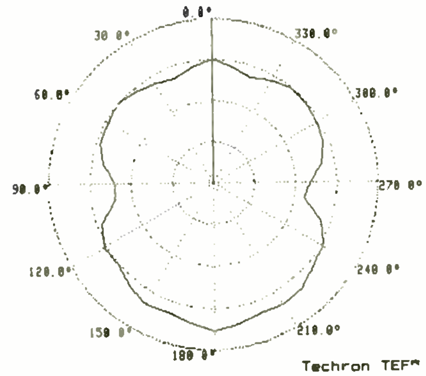



*Figure 10.* ***8 kHz octave band polar response, 10 degrees/data point, 10 dB/ division.***

***"The BES C-70S is unique in its construction and potential uses."***

For those situations where the C-70S may be used in a reverberant space, figure 11 shows its power response. The power response generally follows the axial response. The excessive height of the peak at approximately 19 kHz is due to the averaging process used to generate the power response. This peak is present in all directions and is probably due to a resonance either in the loudspeaker itself or, perhaps, its mounting frame

***"Anyone who is in the professional audio industry and has not heard about the loudspeaker manufactured by BES must have just returned from an extended trip to the moon."***

(see the axial response in figure 3). Figure 12 shows the response as it would be displayed on a one-third octave analyzer. Notice the bump around 200 to 300 Hz.

In listening to this loudspeaker, a definite resonance may be heard that imparts a bit of a "honk" to a male voice. There is also a noticeable lack of low end. A little equalization would certainly help.

The BES C-70S is unique in its construction and potential uses. It looks good and permits rapid and simple installation in suspended ceiling systems. Painted to match a wall, it may be mounted unobtrusively where nothing other than a low profile loudspeaker can be made to look acceptable. Its increased sensitivity off-axis will yield improved coverage in a distributed ceiling system. With a little equalization it should perform well for voice and music.

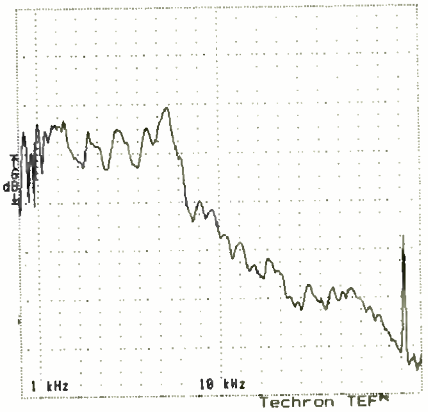



*Figure 11.* ***Power response, 0-20 kHz, 150 Hz resolution.***

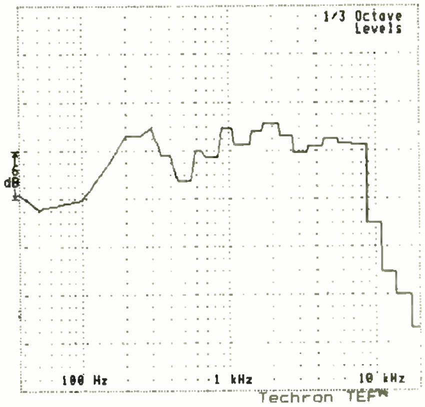



*Figure 12.* ***Power response, 0-20 kHz, ⅓ octave resolution.***

**Farrel M. Becker, a consultant for Audio Artistry, specializes in live sound for the performing arts. Becker started working with TEF technology in 1979 and now teaches the fundamentals of TEF in Techron's training program.**

## BUSINESS FRONT

(continued from page 12)

cultivating what's already growing. It may require learning new technology or methods, seeking new customers and prospects, reeducating old ones. It was hard enough the *last* time, but this time, it's even harder. There is risk associated with it.

The simple fact is, sales reps want these factors accounted for and they want credit, recognition and *money* for pulling it off. Frankly, they deserve all three. One of management's toughest challenges is to create a sales compensation plan that accounts for the multiplicity of variables in the sales environment, the sales personality, and the changing nature of company goals.

***Next Month: Sales Compensation Methods***

*This article was reprinted with permission from the International Communications Industries Association. The article is from the book* How to Compensate the Sales Force *by R.W. Sharer and published by the ICIA. The book can be purchased from the ICIA for $8.50 for members and $11 non-members. Write to: Terri Campbell, ICIA, 3150 Spring St., Fairfax, VA 22031-2399.*

**Bill Sharer is founder and president of Advanced Development Systems, a New Jersey-based firm, consulting in sales, management, marketing communications, and education. For four years, he has been a featured speaker and instructor at NAVA/ICIA Institutes.**

# TEQ™ EQUALIZER TECHNOLOGY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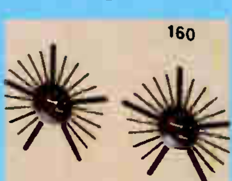
### The benefits of transversal technology . . . Your choice of controls.

With a list of international installations to its credit, and praise from leading acoustical consultants, the TEQ™, Transversal Equalizer, is rapidly becoming the EQ of choice. TEQ™ eliminates unnecessary ringing and transient coloration of signals. While tuned filter equalizers respond correctly at their control frequencies they often create response ripples between control frequencies, causing audible coloration. But the unique TEQ™ design produces frequency responses with a single filter. Its transient response is the weighted response of a single tapped delay chain, and is ripple free over the entire 20dB adjustment range.


TEQ™ with sliders



TEQ™ with screwdriver slots

TEQ™ with rotary knobs

The TEQ™ is now available with the standard ISO 29-band, ⅓ octave frequencies. Also, choose the controls that best fit your particular applications— rotary knobs, sliders or screwdriver slots. Get the benefits of what Don Davis referred to as,

". . . the first genuine step forward in equalizers since the original ⅓ octave units . . ."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## SPEECH SYNTHESIS

*(continued from page 50)*

contrast, is a mathematical representation of the vocal tract as an acoustic waveguide comprising a succession of 12 to 16 uniform tubes with different areas that change dynamically during speech. Digitally represented periodical or aperiodical sound sources (for voiced and unvoiced sounds, respectively) excite the tubes, creating a series of advancing and retreating pressure waves in them. This basic technique has been under much research and development since its inception, and many variations of it have been continually striving towards more realistic synthetic speech, with striking results.

### Dedicated Chips or DSP

During 1977, speech chips began showing up in toys and games. Since then, many computer-based devices, appliances, automobiles, cameras, etc. have incorporated speech synthesis. ICs for both speech synthesis and its recognition are available on both dedicated chip sets and generalized digital signal processing (DSP) chips from many sources. New algorithms are constantly being developed to both model speech and recognize it as well.

### Applications to Speech Systems

One problem that all of us encounter at one time or another is in understanding the spoken word in highly reverberant spaces. Think for a moment of the possibilities afforded to the communications engineer by the following description of a voice recognition and speech synthesis system. An announcement could be encoded into discrete phonemes, then distributed in a digitally encoded format to the appropriate destinations and times. The acoustics of the environment could be considered as part of the vocal tract model and the room delays could be incorporated into the delays found in the vocal tract. Thus the loudspeaker(s) would reproduce an apodized (predistorted) speech signal composed of discrete phonemes, filtered by a system which took into account the filters in the room.

The continuous and transient aspects of connected speech sounds which are normally masked in a highly reverberant environment, and therefore rendered unintelligible, could be electronically produced in a quasi-unconnected state. Remember that the human speech production system superposes one sound upon another due to its inability to rapidly change its structure. These transitional spectra would be produced by the room's "formants". Hence, one could use the room's acoustics to complete the process of producing complete, connected, continuous, and more intelligible speech sounds. Furthermore, if the reverberation of the room is in excess of what the vocal tract model would require, sophisticated signal processing could be used to produce a cancelling signal, thereby minimizing or even negating the destructive reverberant effects.

### Speech Recognition

There are two broad categories of speech recognition: speech to code conversion, and the identification of a person's voice. Our discussion will be limited to the former. Speech recognition is a much more complicated job than speech synthesis because recognition requires a great deal of machine intelligence. However, this can be somewhat simplified depending on the application. There are several types of recognition, including speaker dependent, speaker independent, vocabulary limited, and vocabulary trained. These types can be combined depending on the application. For example, if a system has a half-dozen controllable functions, the computer circuit only needs to learn six words.

Because the components of speech itself are so complex, it takes a lot of computing power simply to recognize the basics, *e.g.*, formant features, vowels, and the many consonant sounds. This is further complicated when we concatenate these sounds into words, and transmit these words as messages. The digitization of speech is rather straightforward and the associated hardware costs are declining rapidly in this area. The preprocessing used may include FFT, formant analysis, feature extraction, and LPC.

Once the speech signal is encoded and preprocessed, some form of pat-





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tern matching takes place. The degree of accuracy in the various schemes is constantly improving, but we have a long way to go. Nevertheless, there are many systems on the market ranging from low-cost systems that can recognize over 30 words, commercially available systems that will recognize up to 1,000 words, to systems now on the drawing board that will recognize up to 20,000 words in real-time. The growth rate in this field of technology is mind boggling when one considers, to add to all this, the combined efforts in artificial intelligence (AI).

Speech intelligibility studies and testing methods have been around for some time now. But the book is far from closed as far as the high-level research world is concerned. Because of the intense activities of both speech synthesis and speech recognition, companies like Texas Instruments are continuing the work started by Bell Labs during the early days of the telephone.

Man has, since the invention of the wheel, been continually devising ways of improving his strength: first, by simple mechanical means; later by machine; and then through electrical and computer technologies. Voice recognition and speech synthesis allow the extension of homosapien to that of robosapien. Robotics and computerization of human/manual tasks are greatly amplified by the application of these new and exciting technologies. The day may come soon when you ask your robosapien to paint the white porch gray—only to be surprised when it returns minutes later with the reply, "The painting is complete. But, that was a red Porsche."


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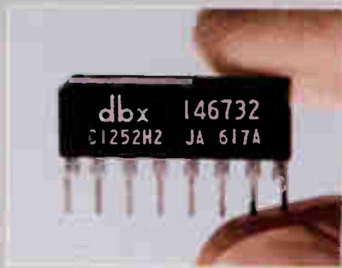
Lawrence R. Rabiner and Ronald W. Schafer, *Digital Processing of Speech Signals*, Prentice-Hall, Englewood Cliffs, N.J., 1978.

# ANNOUNCING THE BIGGEST IDEA IN THE HISTORY OF WIRELESS




## *Samson incorporates dbx* noise reduction to achieve a new standard of wireless performance.*

Samson Concert TD Series is the first true diversity VHF wireless system with dbx noise reduction. We've been setting new standards of sound and reliability in wireless over the years, but we're always refining our technology to make it even better. That's why we chose world-class dbx technology to solve the biggest problem in wireless — noise.

dbx noise reduction makes the Concert TD Series incredibly quiet. It also expands frequency response to deliver the best possible sound for vocals and instruments. And you won't hear it working because dbx circuitry eliminates the pumping and breathing that plagued wireless sound in the past.

Breaking new ground with our Concert TD Series, Samson engineering excellence now takes a quantum leap forward in its ongoing search for the highest standard of wireless performance. We are the first to incorporate dbx noise reduction because we want to guarantee the best possible audio quality available in wireless today. Now there's every reason you can think of to enjoy the unlimited freedom promised by wireless.

First we took the worry out of wireless with reliable no-dropout performance, now we're taking the noise out of wireless with dbx.

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# SERVICES • SUPPORT

## Panduit's Strain Relief Cover For MAS-CON® Connector System

A new strain relief combination cover for use with the MAS-CON connector system has been announced by Panduit Corp.

The optional cover provides additional strain relief in high vibration applications by holding each wire into the slot. The cover can be used with both end and through connectors.

Covers are available for both .100-inch and .156-inch MAS-CON mass terminated connector systems. Covers for .100-inch spacing are available 2 through 28 circuits and for .156-inch spacing, 2 through 24 circuits.

The strain relief covers are molded of natural color 6/6 nylon, same as the connectors.

The MAS-CON connector system is designed for mass termination of discrete wires or flat cable, eliminating the need for hand crimping. It improves termination time and reduces the user's installed cost.


Circle 4 on Reader Response Card

## Fiber Optic Patch Panels Are AT&T ST Series Compatible

A complete line of low loss fiber optic patch panels, connectors, and 62.5/125 micron cable, all 100 percent AT&T ST Series compatible, has been introduced by Atronix, Inc.

Atronix Fiber Optic Patch Panels incorporate up to 10 standard 2.5 mm AMP bayonet connectors into a single 19-inch X 1¾-inch rack mountable assembly. Providing a convenient, low loss method for mating AT&T ST Series or compatible fiber cable assemblies, each patch connector port allows for the attachment of preterminated fibers on both sides of the panel.

Featuring bayonet jack connectors, typical loss characteristics for a properly terminated mated connector pair on the Atronix Fiber Optic Patch Panel is less than 1.0 dB. Preterminated and 100 percent optically tested fiber optic cable assemblies including 62.5/125 micron fibers with single, duplex, and up to 24 optic fibers in a single, jacketed cable assembly are also available.


Circle 5 on Reader Response Card

## Progressive Electronics' Load Coil Detector

Progressive Electronics, Inc. has announced the introduction of the Model 88, Load Coil Detector. Telco's can economically equip outside personnel to test lines for the presence and number of load coils. It saves field personnel time; no waiting for Central Office people to test the line. Selling for $65, the Model 88 is small (four-inch X two and one-quarter-inch X one and one-half-inch), light weight (four ounces), and easy to operate (one control knob, a meter, and two test leads). This unit is housed in a high impact plastic case.

Circle 6 on Reader Response Card

## Color Jacketed Computer Cables Simplify Identification

Belden Wire and Cable now offers Belden 9269, a RG-62 A/U-type coaxial computer cable with outer jacket colors of orange, yellow, light blue, chrome, and beige, in addition to black. Suitable for office environments, these color-coded cables ease circuit identification, simplifying the internal and external connection of computer terminals, according to the company.

Belden 9269 is a 93-ohm coaxial cable with polyethylene insulated 22 AWG bare copper covered steel conductor and a copper braid shield that provides 95-percent coverage. The cable has a nominal capacitance of 13.5 pF/feet and a velocity of propogation of 84 percent.

Belden 9269 is available in Un Reel® packaging for easy installation.

*Circle 7 on Reader Response Card*

### Viking Acoustics' PC Work Station

Viking Acoustical Corportion has released a PC workstation that combines its two popular product lines: acoustic printer enclosures and ergonomic computer furniture.

The Quiet Station is a standalone workstation that takes up five square feet of floor space, while it holds and positions computer and peripherals. The new Quiet Station also features a built-in silencer that houses a computer printer.

Other features include: two pull-up platforms snap in place for extra work-surface; slide-out keyboard tray at optimum ergonomic height; monitor is positioned for most comfortable view-

ing angle; lockable monitor cabinet fits most PCs including IBM AT; silencer top slides out for convenient paper loading and fits most popular 80 and 132-column printers; rear-access door for each access to cabling; software storage area; available with optional slide-out file drawers.

The quiet station is available in teak, oak, walnut and putty finishes.

*Circle 8 on Reader Response Card*

### Line Boosters Double Allowable Cable Length

The Black Box® 232 Line Booster receives and retransmits the 12 most commonly used signals of the RS-232C interface, resulting in doubling the signal transmission distance beyond the accepted RS-232 specification of 50 feet. Data rate and data format transparent, this device can be powered either from the interface or a detachable power supply.

# Ad Index



The Centronics Line Booster does all the above, except it accommodates the Centronics® compatible printer interface. The 16 most common signal lines of this interface are supported and the device enables doubling of the cable length (10-15 feet) typically permitted in a TTL-driven application.

Either type device ($99) is installed at the center of the RS-232 or Centronics cable.

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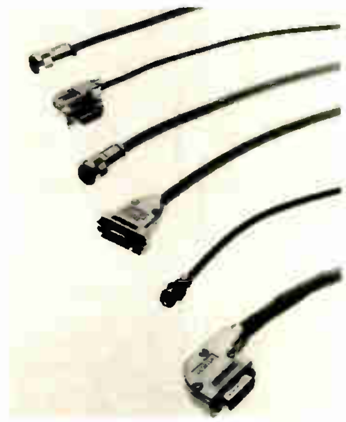


**Ethernet Cable Assemblies Meet IEEE 802.3 Specifications**

A complete line of Ethernet® cables and accessories for DEC, Xerox, Ungerman-Bass, Interlan, and other local area networks is being introduced by Atronix, Inc.

Atronix Ethernet Products feature OEM specified baseband coaxial and transceiver cables, Thin and Ethernet 50 ohm miniature coaxial cable, and this transceiver cable, as specified by DEC for office drops. Available in PVC or Teflon®, they are manufactured to IEEE 802.3 specifications.

Transceiver cables come in 180° or 90° terminations with either polyethylene inner conductor insulation or with Teflon® for use in air return plenums. Atronix Ethernet Products can be supplied in bulk or as complete assemblies with connectors and hardware.

*Circle 10 on Reader Response Card*

**New Winsted Equipment Rack Holds Over 1000 Pounds**

A new heavy duty equipment rack from the Winsted Corporation is constructed to accommodate more than 1000 pounds of electronic equipment. The (Model V8701) 78-3/4 inches industrial quality rack provides 30 inches of depth with rack space for mounting of electronic equipment.

*Circle 11 on Reader Response Card*

# PRODUCTS IN REVIEW

## PASO INTRODUCES MICROPHONE ACCESSORIES

Paso has announced the introduction of a new line of satin black microphone floor stands, desk stands, booms and goosenecks.

Floor stand models include the B-10B light duty tripod stand and the B44B heavy duty stand with collapsible die cast base. The B44B's design incorporates the use of synthetic rubber "feet" which isolate the stand and the microphone from vibrations that can cause feedback. To adjust the height, both stands utilize a nylon clutch mechanism which insures reliability.

The A-13B boom provides an easy to use "quick set" clamp which allows for spontaneous adjustments. The integral and compact counterweight guarantees stability when positioning. The A-13B may be used with either the B10B or B44B floor stand.

The B-16B may be used as a desk or floor stand because of it's adjustable height between 16 and 24 inches. The heavy five pound die cast wide stance base prevents "tipping." This stand may be used in broadcast, studio recording or on the road. A felt cover protects conference tables from scratching. Completing the lineup of microphone accessories is the GNK15B 15 inch noiseless black gooseneck. The non reflective black anodized finish eliminates glare and reduces the obtrusive appearance. Standard five-eighths 27 tread for mounting.

The B60B black gooseneck flange facilitates surface mounting. This line of stands and accessories is also available in chrome.

*Circle 12 on Reader Response Card*

## TEKTONE'S DOCTOR'S IN/OUT REGISTER

TekTone's DR series IN/OUT Register Systems provide visual indication of the presence or absence of an individual through the use of the system's master register and slave annunciator panels. Message panels are also available for a complete health care facility or office communication system.

Visual signaling is indicated by high intensity red LED's and controlled by selective, push-push type switches corresponding to a particular name on the register. All components are mounted on a printed circuit board.

Panels are modular in design and constructed of clear, anodized aluminum. Master and slave register panels include an engraveable, easily accessible name directory, protected by a vandal-resistant window.

*Circle 13 on Reader Response Card*

## SMC INTRODUCES AUDIO TUNER

The NT-1 is a fixed frequency, crystal controlled, computer generated, superhetrodyne tuner. The front end of the NT-1 is crystal controlled and uses a diode protected dual gate D-MOS field effect RF amplifier with a noise figure of 1.5 dB or better at 100 MHz. This device is capable of 55 dB liner gain reduction versus typical 30 to 35 dB in earlier MOS FET type circuits resulting in an overall dynamic range of over 100 dB. The input and output impedances remain constant, preventing any detuning of the high "Q" RF circuits. Three high "Q" tuned circuits and a dual gate MOS-FET mixer greatly enhances the dynamic range over conventional bi-polar mixers, according to the company.

A newer, more state of the art IF system has been included which eliminates the multisection 10.7 MHz IF band-pass filter. Since crosstalk and selectivity are effected by the characteristics of earlier design L-C filters, the NT-1 provides better selectivity to reject unwanted signals from alternate channels, and crosstalk is excellent.

The complete subchannel system is incorporated in a specially designed new monolithic chip, which provides 60 dB of limiting and a special demodulator permitting exceptionally good AM rejection and signal-to-noise ratio.

A specially designed mute circuit is incorporated to provide noise free muting regardless of the time constant of the SCA generator in the transmitter. The mute circuit can also be adjusted to mute when the signal-to-noise deteriorates below the desired level. SCA muting is derived from a dual sensing circuit which senses the wide band noise of the main channel rather than the RF input level and the absence and presence of the SCA carrier.

The subchannel audio output is available on the rear panel through a phono jack at a level of one volt. A built in isolation transformer has been included to provide a balanced 600 ohm output. This eliminates the need for an optional output transformer.

The number of components have been greatly reduced due to the use of the new main channel IF system and the single integrated circuit for the complete SCA and audio system; fewer components means greater reliability.

The internal circuitry is designed to operate at approximately 13 volts DC. This voltage is derived from the internally mounted 120 V AC high reluctance transformer and bridge rectifier configuration. A three pronged AC cord is used to insure that the NT-1 is properly grounded.

Power is available for driving an external RF amplifier by simply relocating the antenna solderless input lead on the PC board from terminal "A" to "B". A rack panel is available for rack mounting the NT-1 Tuner.

*Circle 14 on Reader Response Card*

## PROFESSIONAL AUDIO INTRODUCES PI SYSTEMS

Professional Audio Systems has introduced its PI (Permanent Installations) Systems.

The PAS PI-152 is designed for a wide variety of sound system applications, as a stand-alone device, or with the PI-181 subwoofer system. Based upon the CX-1580C fifteen-inch CoAxial loudspeaker, it features excellent sensitivity, high power capacity, and smooth, extended low and high frequency response, according to the company. The high frequency horn is of the constant coverage type, and provides uniform 30 by 60 degree distribution to beyond the audible frequency range.

The PAS PI-122 is designed for a wide variety of sound reinforcement and playback applications, either as a

*(continued on page 72)*

[redacted]

(continued from page 70)
stand-alone device, or as a modular component of larger systems. Based upon the XC-1280C 12-inch CoAxial loudspeaker, it features high sensitivity, robust power capacity and phase-coherent extended response. The high frequency horn is of the costant coverage type, providing uniform 30 by 60 degree distribution to beyond the audible frequency range.

The PI-122P includes a passive frequency dividing network featuring Time Offset Correction (TOC™). The PI-122A is intended for biamplification, and is recommended for use with the PI-1 system processor. The PI-122A is shipped with a high frequency driver DC blocking capacitor installed.

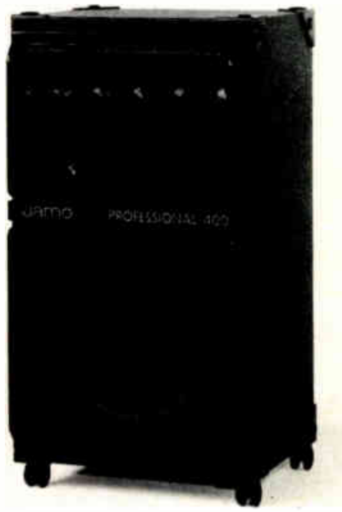
The PI-181 is a subwoofer system that is designed to operate from 30 Hz to 100 Hz, and is an ideal complement to the PI-122 and PI-152 CoAxial systems. Housing a single PAS ER-1880C 18 inch loudspeaker in an optimally-ported direct radiator enclosure, the PI-181 features excellent sensitivity, high power capacity and low distortion.

The PI-1 is a line-level dividing network and system processor for use with PAS PI Series loudspeaker systems in biamplified and triamplified applications. The PI-1 includes two-way and three-way active crossover circuitry with constant group delay, Time Offset Correction (TOC™).

**Circle 15 on Reader Response Card**

## JAMO'S PROFESSIONAL SERIES 400 LOUDSPEAKERS

Handling 560W peaks live or recorded, the Professional 400 also stacks up to any other monitor on the market thanks to its self-locking fixtures, standard on all three JAMO Professional Series systems. The 400 has five horn-mounted tweeters in a proprietary Controlled Dispersion™ array for highs the full equal of its 15-inch woofer lows. Its precision-engineered anti-turbulence cone at the throat of the midrange horn further



increases linearity and reduces distortion. The result is a system for professional installations that sacrifices none of its musical integrity to its outstanding and obvious ruggedness according to the company. Weighing in at 92.6 pounds, the Professional 400 has a suggested retail price of $899.95.

**Circle 16 on Reader Response Card**



**Circle 233 on Reader Response Card**

## AUDIO CONTROL INDUSTRIAL'S ANALYZER

Audio Control Industrial has announced it is now shipping the first affordable, fully professional, third octave spectrum analyzer, real time program monitor with memory. This model, the SA-3050, meets or exceeds ANSI Class II standards utilizing fourth order filters for accuracy.

The analyzer package includes a calibrated microphone for the price of $850.

For recording and broadcast, applications include the ability to monitor spectral balance during recording, overdubbing and mixdown. Accurate measurement for correction of studio monitor speakers is another important use.

The 30-band display uses an easy-to-read, large format 270 dot matrix grid. Controls for this display include resolution of one, two, three, and four dB per step, SPL on-off, and fast, medium and slow decay integration. The total available display range is 44 dBa to 136 dBa or −56 to +36 dBm.

Other standard features are six memories with read, write, freeze, and RTA-memory comparison functions. Three signal inputs are standard; a phantom powered balanced XLR input, an FET instrumentation BNC input, and a balanced bridging ¼ inch line level input. The SA-3050 includes a precision digital pink noise generator

with adjustable output that has the power to drive a speaker directly. Also included in the package is a calibrated phantom powered ¼ inch pressure gradient condenser measurement microphone.

Options which further enhance the analyzer's usefulness also are available.

An internal rechargable battery package is one which permits portable operation. A parallel Centronics-type printer interface is another option available so that a common, inexpensive printer may be driven directly. An expansion port for future applications and options is provided as well. A rack mount frame and soft carrying case are also optional.

**Circle 17 on Reader Response Card**

## ROWE INTRODUCES ENTERTAINMENT CENTER

Rowe's Entertainment Center, is a self contained record and video tape player that looks like a Rowe top-of-the-line jukebox, but allows customers to make their own selection at no cost.

The center is connected to an existing speaker system or to remote video monitors—including wide screen or projection.

The center is 54 inches high, 42 inches wide and 27 inches deep.

**Circle 18 on Reader Response Card**

## DSP CHIP

*(continued from page 53)*

Audio preprocessing such as preamplification and filtering must be combined with multiplexing and analog-digital conversion to convert the signal into digital format. Extra memory, logic "glue" devices, clock circuits, control panel interface components, and perhaps a conventional microprocessor for "housekeeping" are all needed to support and complement the functions of the DSP chip. Digital-analog and post-filtering analog circuitry (at the output of the system) and power supplies are among the other support devices that comprise a DSP audio product.

**Application-Specific IC's**

The "one chip" digital signal processing product is still a few years away. Nonetheless, advances in the field of custom or "application specific" integrated circuit (ASIC) development have already made it possible to design a custom IC which includes a DSP section similar to the one described for high volume products.

The custom chip is designed using computer-aided design and simulation techniques which reduce the number of costly fabrication and debugging steps involved with producing chip-level products. Additionally, the use of standardized chip-level "components" and "Silicon Compiler" systems is bringing the almost unimaginably complex task of designing an IC with hundreds of thousands of components down to a level where a single engineer-programmer can handle the task.

Approaching a product design from the chip level, as opposed to designing a system composed of standard parts, is definitely the wave of the near future. This is critical to the introduction of products with true mass appeal and consumer-level pricing.

Digital signal processing implemented in these new types of devices will lead the way to more powerful, more easily changed, managed, and built audio signal control systems. Filters, spectrum analyzers, reverb and special effects, mixing consoles, smart signal processors, and special function controllers all are being developed and are now starting to enter the marketplace.

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**Michael Beigel, who has been a consultant since 1978, specializes in electronic product development and holds numerous patents.**

# PRODUCTS IN REVIEW

## a closer look

by gary d. davis

**Barcus-Berry Electronics**
**BBE 802 Signal Processor**

Barcus-Berry Electronics, Inc. has announced the availability of a new, lower cost model of its signal processor, aimed at the professional and commercial sound marketplace.

Called the BBE 802 signal processor, this pro sound model has a suggested professional net price of $499, according to John C. McLaren, chairman of Barcus-Berry Electronics.

For use in recording studios, live concerts, TV and radio broadcasting and other commercial sound applications, the BBE 802 is a multi-band, program-controlled signal processor which can improve the overall sonic quality of virtually any reproduced sound.

The BBE 802 uses high-speed dynamic gain-control circuitry to audibly improve the reproduction of program transients, adding brightness and presence without introducing the undesirable stridency so often characteristic of equalized sound, according to McLaren. It increases voice intelligibility by eliminating frequency-band masking when important sibilant and consonant elements are represented in the program signal.

The dual-channel BBE 802 is packaged in a rack-mounted chassis which occupies one standard EIA space. Its dimensions are 17-inches wide, 1.75-inches high, and eight-inches deep. Easily installed in any audio chain, it required nothing more than insertion between the program source and an amplifier, recorder or signal transmission line.

McLaren said, "All of us at Barcus-Berry Electronics are tremendously pleased with the industry's acceptance of our patented BBE process. It's especially gratifying to hear the experts mention the BBE process in the same breath with stereo, FM, Dolby and compact discs as major breakthroughs in audio technology," McLaren added.

With the pro sound BBE 802, phase adjustments are primarily directed toward preventing high-frequency time lag, or transient distortion, and the automatic gain changes are based on interband program amplitude ratios. Swept frequency response of the system is essentially flat from 20 Hz to 20 kHz in the bypass mode. In the process mode, high-frequency response is controlled by program. Amplitude changes are developed only in direct response to application of a spectrally-diverse program signal.

**Comments:** Okay, how many of you completely understand what the BBE 802 does based on the information above, provided by the company? I sure didn't, so I called BBE and spoke with engineer Ryan Coutts. This signal processor is designed essentially to pre-distort the sound going to the loudspeakers to correct for the kinds of phase and harmonic amplitude errors typically created by loudspeakers. Simple. But how does it work?

You have probably read about, or used, power amplifiers that include some sort of servo circuitry (feedback from the speaker output more closely resembles the amp's input signal. The back EMF from the speaker can convey a lot of useful information in this regard. The problem is that this method is (a) expensive, (b) only works if installed on a given sound system, and (c) is tricky to implement in large, multi-amped sound systems. Barcus-Berry used a servo circuit to do their research and found a number of similarities in the way that most loudspeakers distorted the sound. Consequently, they were able to develop a pre-distortion processor (the BBE 802) which goes between the preamp or mixer and any power amplifier. In fact, the processing can be applied to a recording (and is already being used by at least one major mastering facility in Los Angeles). When applied to recorded or broadcast programs, the processing enables the listener to hear the benefits without purchasing any special equipment.

The processing itself has two major aspects: phase shift and harmonic companding. The phase shift is applied in a fixed, pre-determined manner modeled according to Barcus-Berry research. The companding "looks at" signal in two bands; mid-range sound is considered to be primarily "fundamentals" and higher frequency sound above 1.25 kHz is considered to be "upper harmonics." The circuitry compresses or expands frequencies above 1.25 kHz by up to 11 dB; expansion is based on the "harmonics" and compression on the "fundamentals." The operator can adjust the ratio of upper harmonics with a front panel control.

There is another circuit in the BBE 802 that serves as a low frequency filter, permitting the user to crank in up to 7 dB or so of boost below 250 Hz using a recessed, front-panel, screwdriver-adjustable control. We get the feeling this was kind of an add-on, not central to the processing, and really could be done with a conventional EQ or filter set.

Designed for professional interface, the two-channel BBE 802 sports balanced differential XLR inputs, floating transformer-isolated XLR outputs, +4 dBu nominal level, +24 dBu maximum level, output noise below −90 dBu, and THD of 0.5% at 1 kHz, +4 dBu. It is nominally a unity gain device, but does have an input level control.

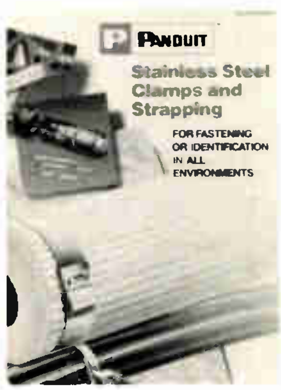
The BBE 802 is intriguing in its concept. It is definitely intended for live or recorded sound, but not as an "effect" since the mixing of direct with processed sound in an effects loop would cause unwanted cancellation due to the phase shift circuitry. The proof, as they say, is in the pudding, so we feel the BBE 802 deserves your *closer look.*

*Circle 19 on Reader Response Card*

## Panduit® Bulletin Describes Stainless Steel Clamps

A free newly revised full color bulletin is available from Panduit Corp. It describes the company's line of stainless steel clamps and strapping for applications involving abrasion, weathering, corrosion, radiation or temperature extremes. They can be used for fastening or identification indoors, outdoors and underground. The 12 page bulletin provides detailed descriptions of the two systems, including material specifications.

Stainless steel clamps are available in either Type 302/304 or Type 316 stainless, with 316 recommended for applications requiring higher corrosion resistance. Stock sizes are furnished in standard cross section for bundle diameter up to eight inches or heavy cross section for bundle diameter up to 10 inches. Minimum loop tensile strength is 100 pounds and 250 pounds respectively. For larger diameters or job site versatility of any diameter, Panduit offers heavy cross section banding in 1,000 foot




reels, with heads furnished separately.

Strapping is available in Type 302/304 stainless in three widths for bundle diameters up to 10 inches. It is also available in 200 foot reels for larger diameter applications.

Installation tooling for both systems is also included, along with accessories and a new metal embossed tape system.

*Circle 1 on Reader Response Card*

## Contact East Offers Free Source Book

Contact East's product guide for 1987 features a full-line of tools and instruments including hundreds of new products. Highlighted are the latest in static control, an expanded tool kit line and state-of-the-art fiber optic equipment. Their product line emphasizes test instruments (DMM's, oscilloscopes, counters, power line monitors, disk drive testers), hand tools (screwdrivers, pliers, wrenches, crimpers) soldering supplies and Contact East's exclusive line of tool kits for any application. All products are described in detail with specifications, full color photos, prices and are 100 percent guaranteed. Every order, by phone or mail, placed by three p.m.(est) is shipped from their warehouse, by five p.m.(est).

*Circle 2 on Reader Response Card*

## Belden Releases Innovators

A 42-page booklet from Belden entitled *Innovators* has been released. The latest issue contains articles on fiber optics, conformable shields and swept impedance measurements. The articles are illustrated with pictures and charts.

*Circle 3 on Reader Response Card*

# FACES AND PLACES

## Com Dev Appoints Groome Dir. of Market Development

Preston Groome has been named director of market development at Com Dev. The former regional sales manager for the Northeast Region will now concentrate on identification and development of end user markets for Com Dev products. Groome joined Com Dev in 1979.

In his new capacity, Groome's initial efforts will be directed at national accounts, Fortune 500 companies, and other large multi-location companies. Other groups, such as Federal agencies, AMTRAK, Freddie Max, as well as state, county, and city governments, that are authorized to purchase under GSA contracts are included. Long distance resellers and international markets are also targeted.

According to Richard Mugavero, president, "Preston's new responsibilities support our strategic plan to achieve, within two years, 15 percent to 20 percent of our sales to these targeted groups."

Charlie Cook, previously communiations manager for Merrill Lynch, has joined Com Dev to fill the Northeast region sales manager position.

STEVE ROMEO

## JBL Hires Romeo As Market Manager

Steve G. Romeo has been hired as market manager at JBL Professional, overseeing the sound contracting, motion picture and broadcast markets.

At JBL, Romeo will be responsible for new product development as well as field support of dealers, sound contractors and consultants. In addition, he will act as project manager for JBL's Central Array Design Program, a computer software system that aids sound contractors in the design of sound systems.

Prior to joining JBL, Romeo served as manager of market development for both domestic and international markets for the Bose Corporation of Framingham, Massachusetts. He has been involved in live sound reinforcement and sound contracting since 1973.

A graduate of the University of Colorado, Boulder, Colorado, Romeo resides in Van Nuys, California.

JOHN R. CHARCZUK

## Jeron Appoints Charczuk Vice President, Sales

The appointment of John R. Charczuk to the position of vice president, sales has been announced by Jeron Electronic Systems, Inc.

# REP NEWS

**Video Data Systems** has announced the appointment of **Mega Hertz Sales** as their representative in Texas and the Rocky Mountain and Central Plains States. Mega Hertz Sales has offices in Denver, Dallas and Saint Louis. Video Data Systems also announced that **MPCS Video Industries** has been appointed as their dealer in the New York Metro Area and that **Omnivue, Inc.** has been appointed their representative in the Northeast.

Aiphone Corporation has announced that **B.C. Electronics**, a commercial sound and security firm based in Mission, Kansas has been appointed their manufacturer's representative in five western states. B.C. Electronics, which has an office in Saint Louis will cover Nebraska, Kansas, Missouri, Iowa and southern Illinois.

**Frazier** has announced the appointment of the following firms to represent its loudspeaker products: **Dimension Marketing** for Texas, Louisiana, Arkansas, and Oklahoma; **ELREP Sales** for the Southeast; **Vector Corporation** for the Pacific Northwest; **Alby Currant Sales** for Upstate New York and **J. C. Electronics** for Hawaii.

CALENDAR OF EVENTS

# DATEBOOK

| DATE | EVENT/COMMENT | LOCATION | CONTACT |
| --- | --- | --- | --- |
| March 24-26 | **"Telephone Technology and Practice."** Workshop offered by abc Teletraining, Inc. | Phoenix, AZ | abc Teletraining (312) 879-9000 |
| March 31-April 2 | **Automated Design and Engineering for Electronics West.** | Anaheim, CA | Cahners Exposition Center (312) 299-9311 |
| April 2-3 | **Understanding ISDN.** | Las Vegas,NV | Telecommunications Research In Associates (312) 553-ISDN |
| April 5-8 | **NSCA Contractor's Expo '87.** Exhibits and workshops. | New Orleans, LA | NSCA (312) 593-8360 |
| May 4-5 | **Computers and Communications in the Health Industry.** Conference sponsored by Frost and Sullivan. | Dallas, TX | Frost and Sullivan (212) 233-1080 |
| May 5-7 | **1987 Static Overstress Seminars.** | Phoenix/Scottsdale, AZ | Judy Ward 1-800-826-6270 |
| May 13-15 | **EDS '87.** National forum for electronic distribution. | Las Vegas, NV | EDS (312) 648-1140 |
| June 8-10 | **Federation of Conferences of Information Systems.** | Washington, D.C. | The Federation (617) 358-5356 |
| June 8-11 | **"Instrumentation for Engineering Measurements."** Course presented by the Center for Professional Advancement. | East Brunswick, NJ | The Center (201) 238-1600 |
| June 12-14 | **Recording Studio Designer's Workshop.** Sponsored by Syn-Aud-Con. | Astoria, NY | Syn-Aud-Con (812) 275-3853 |
| June 15-16 | **CAE.** A conference sponsored by Frost and Sullivan. | Boston, MA | Frost and Sullivan (212) 233-1080. |
| June 15-18 | **"Discover the Power of Information."** National Computer Conference. | Chicago, IL | NCC '87 1-800-NCC-1987 |


# GET THE FACTS

## Watch NSCA-TV News for complete coverage of Expo '87

NSCA-TV WILL BE ON-THE-AIR, EVERY HOUR, IN EVERY HOTEL ROOM AT THE FAIRMONT, LE PAVILLON, MONTELEONE, HOWARD JOHNSONS AND ON LARGE-SCREEN TELEVISIONS ON THE EXHIBIT FLOOR for all three days of the NSCA Contractor's Expo '87. NSCA-TV News will provide convention goers with up-to-the-minute, immediate information on exhibitor's new products, programs and trends.

NSCA-TV is written & edited by *Sound & Communications*. It is produced by Testa Communications and in no way is associated directly with the National Sound & Communications Association.

# JBL Helps Morgan Sound Meet Deadlines

To meet some of the tightest deadlines in the history of concert sound reinforcement, Morgan Sound, Inc. has made several purchases of direct-from-stock JBL Concert Series equipment.

"Our initial purchase orders, worth more than $150,000 were entered four weeks prior to our first show date at the Yellowstone County Fair in Billings, Montana last August," said Morgan Sound president Charlie Morgan. "We were able to do something that's very rare in the concert sound field—sign contracts with the promoters before receiving our hanging cabinetry."

Among the Concert Series equipment that was installed in the 15,000 seat venue were 36 model 4852 two-way enclosures and 12 model 4842 subwoofer assemblies. In addition 33 UREI model 6290 power amps and a UREI model 525 electronic crossover were used.

To accommodate the varied sound reinforcement needs of fair headliners, Morgan Sound supervised the creation of four trapezoidal bars, each holding nine speaker boxes. Bars were fabricated by Rick Reed Productions.

Morgan Sound used the same Concert Series configuration to provide sound for Seattle's Bumbershoot Festival, a Labor Day weekend celebration.

## Caraway Audio Provides Sound for Great America

Caraway Audio provided sound reinforcement for Great America when it unveiled its new entertainment venue called the Redwood Amphitheatre which seats 10,000 and features detailed woodwork throughout the facility.

"We brought in 16 stacks of 32 Harbingers bins for the main system. In this package we got two inch beryllium top end, Mark Wayne/E.A.R. custom designed 12 inch mid cones and 18 inch Cerwin Vega subs," said Doug Caraway the audio firm's owner. "All the power was provided by the E.A.R. M 2000 MOSFET Power System. We have incredible results everytime we use this top-line sound package."

# PEIRCE-PHELPS JOINS THE ACT

The Audio Systems Division of Peirce-Phelps, Inc. has completed installation of a sound system it designed for the experimental "black box" Tustin Theater at Bucknell University, Lewisburg, PA

The theater is essentially a black box with four walls, bare floor and lighting grid which allows for maximum flexibility in putting on productions, explained Steve Chene, lighting designer and associate technical director for the university's Theater Department.

An advantage of the avant garde concept is that the stage can be placed anywhere, according to Chene. "We can position the stage in a three-quarter thrust, conventionally, or mix actors and audience together." The theater can accommodate 250 people with different seating configurations and measures 50-feet square by 18-feet high.

Chene said the sound system plays an integral role in staging. The system, for example, features a ceiling grid of 16 Electro-Voice PI-100 speakers that can be physically or electronically aligned to develop point sources depending on production requirements. "Sound always sounds like it is coming from the stage area, regardless of where we place it. Each speaker also has its own 100-watt amplifier and we have the capability of doing some incredible sound effects," Chene said.

Other components of the system are a Soundcraft 200B 16-channel mixing console and an Audio-Technica Unipoint microphone system.

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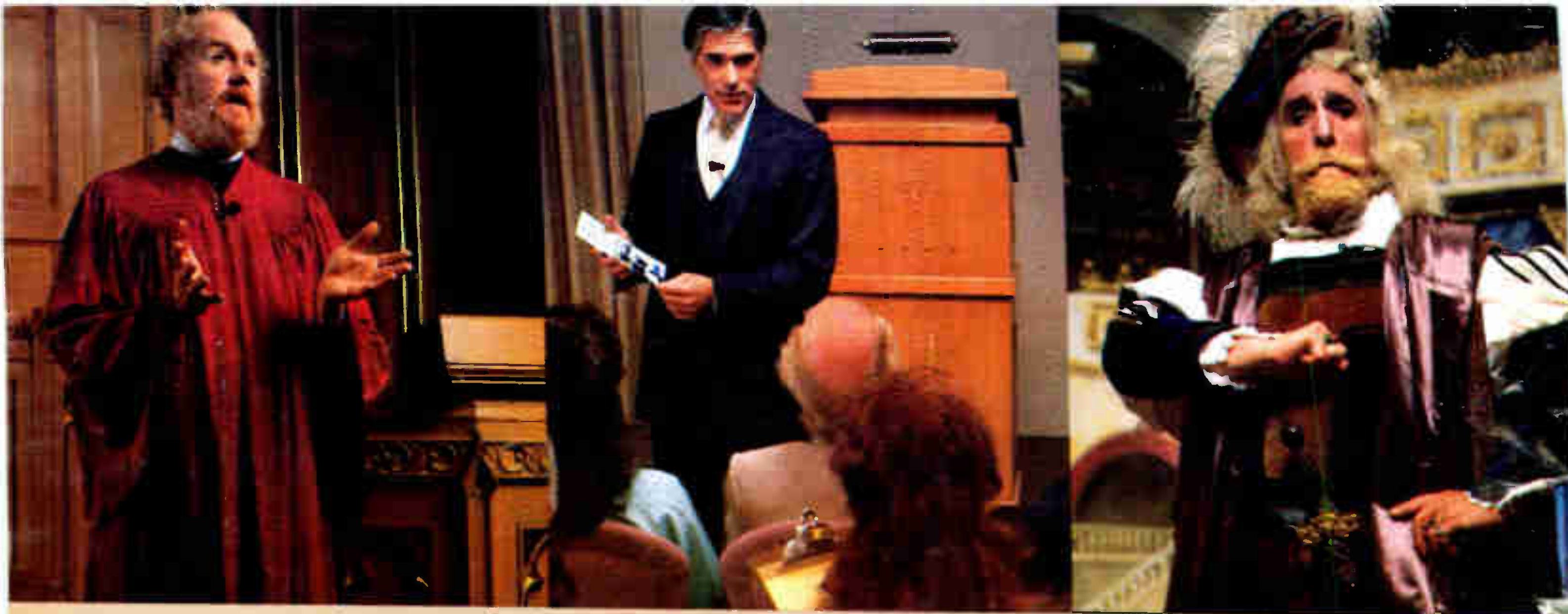

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**THE SHURE WIRELESS**