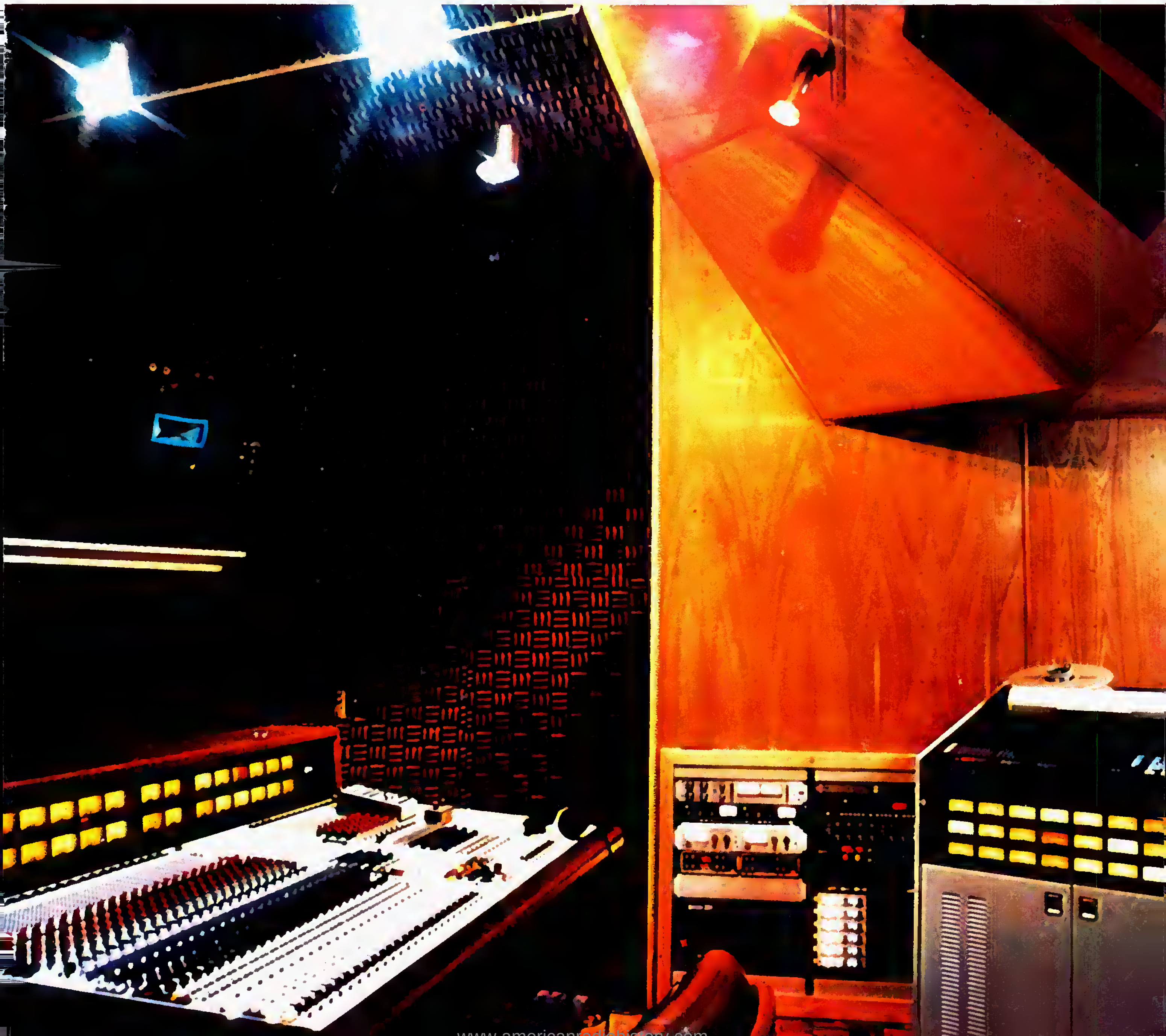
THE SOUND ENGINEERING MAGAZINE

db

FEBRUARY 1983
$1.95



"No noise, nor silence, but one equal music."

John Donne, 1571-1631.

The new Klark-Teknik high-performance DN30/30 graphic equaliser offers much more than just a quiet ability to balance channels right across the audio spectrum. Thoughtful ergonomics are backed by a new circuit design breakthrough using ultra-stable microelectronic filter networks to set performance standards comparable with Klark-Teknik's 'golden oldie' the DN27A. The DN30/30 is the equaliser to boost a studio's reputation, meet broadcasting specs in less rackspace, cut costs and equipment failures on the road — because . . .

It fits two matched high specification graphic channels into a single unit, each providing ⅓ octave equalisation over a full 30 ISO centre frequencies.

It gives fine fingertip low-frequency control covering the subwoofer range down to 25Hz — with touch-sensed centre detents, selectable cut boost level range and fail-safe design giving extra certainty during live events.

Its advanced design, tough construction, stringent testing and long burn-in exceed even Klark-Teknik's previously high standards for reliability and consistent performance on the road.

For technical information ask for:
- Our DN60/RT60 Data Sheet.
- Our DN30/30 Data Sheet.
- Our Application Notes on equalisation.

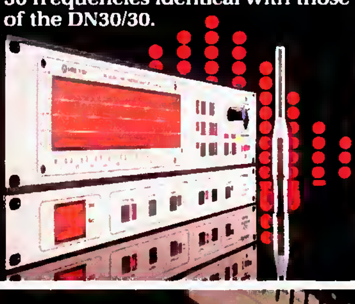
KLARK TEKNIK
sound science

Klark-Teknik Electronics Inc.
262a Eastern Parkway, Farmingdale, NY 11735, USA.
Telephone: (516) 249-3660

Omnimedia Corporation Limited
9653 Côte de Liesse/Dorval, Quebec H9P 1A3, Canada.
Telephone: (514) 636 9971

Manufactured by Klark-Teknik Research Limited, England.

SOME MORE EQUAL THAN OTHERS
The microprocessor-based DN60 Spectrum Analyser and RT60 Reverberation Analysers provide LED matrix display of system, performer and studio response at 30 frequencies identical with those of the DN30/30.



Circle No. 11 on Reader Service Card

db

FEBRUARY 1983 VOLUME 17, NO. 2

FEATURES

DEPARTMENTS

About The Cover

• This month's cover spotlights the LEDE Control Room at Telearte Recording Studio in Caracas, Venezuela. The audio equipment featured in the control room includes a Sphere Eclipse Console, Sonex, an Ampex ATR-124 multitrack recorder, UREI limiters and Time-Aligned monitors, a Lexicon Prime Time Digital Delay, an Eventide H949 Harmonizer, and Valley People processors. For more information, see Sherman Keene's article on page 38.

db, the Sound Engineering Magazine (ISSN 0011-7145) is published monthly by Sagamore Publishing Company, Inc. Entire contents copyright © 1983 by Sagamore Publishing Co., 1120 Old Country Road, Plainview, L.I., N.Y. 11803. Telephone (516) 433 6530. db is published for those individuals and firms in professional audio-recording, broadcast, audio-visual, sound reinforcement, consultants, video recording, film sound, etc. Application should be made on the subscription form in the rear of each issue. Subscriptions are $15.00 per year ($28.00 per year outside U.S. Possessions; $16.00 per year Canada) in U.S. funds. Single copies are $1.95 each. Editorial, Publishing and Sales Offices: 1120 Old Country Road, Plainview, New York 11803. Controlled circulation postage paid at Plainview, NY 11803 and an additional mailing office.

Index of Advertisers

Coming Next Month

• Next month, in conjunction with the upcoming NAB Convention, will be our Broadcast issue. Among the featured stories will be a piece by Joel Tall and Skip Pizzi on recording via telephone, and an article on sound mixing for video, the Editel way, by Vin Gizzi. Also, Vladimir Nikanorov explains the various means of transmitting audio to and from satellites using the latest digital methods, while Bob Metzler of Tektronix brings us a look at Automated Test Systems. In addition, Ken Pohlmann reviews the MCI JH-800 broadcast console. Of course, our regular columnists and departments will also be on hand. All this—and more—coming in March's **db—The Sound Engineering Magazine.**

Letters

CONTRACTION REACTION

To The Editor:

I noted with interest Mr. Kirkendall's letter and your reply in the September 1982 issue. Your statements in regard to db need no further comment. However, the word "microphoning" has also been one that I have struggled with. I have resolved the dilemma for myself by the following rationale: "Mic" is an abbreviation of microphone, which due to its common use, does not need a period. "Mic'ing" is a contraction, the apostrophe giving a clear indication that rodents are not being discussed.

Though I am far from an expert on matters of editorial style, the above does not appear to be in conflict with style manuals such as that of the American Institute of Physics (used by the Acoustical Society of America, a member body of AIP), except for the general prohibition against the use of contractions in formal papers, scientific journals and the like.

In closing, all I can say is that I'm glad that I'm involved with the input end of the audio chain...what do the other guys do to solve the problem of "loudspeakering technique???"

Geoffrey M. Langdon
Sennheiser Electronic Corporation

db replies:

Now that you mention it, we're not sure what the loudspeaker folks do. But next time we're 'speak'ing to one of them, we'll ask.

VEXED OVER HEX

To The Editor:

Wouldn't it be easier to read digital (binary) numbers if they were separated by commas after every fourth digit? For example: 1100,1001,0001,1000. This makes for easy hex conversion, and without further explanation identifies the number as having a binary base.

D. Fish
Golden Triangle Recording, Inc.
Denton, Texas

db replies:

The commas certainly make it easier to read than 1100100100011000. However, base-2 numbers aren't often seen in print in lengths longer than eight bits (11001001, for example). When it is necessary to print a longer bit stream, a space is often used in place of a comma, as in 1100 1001 0001 1000. This is also a lot easier to understand than an endless stream of 1s and 0s.

V8 WE'RE NOT

TO THE EDITOR:

After getting dulled to the bone by computer and electronic's "experts" trying to supply me with digital technical info, I slapped my head and exclaimed: "Wow! I coulda had a **db**!" You guys are the best, and I know you'll get me what I need. Being a music person (but non-engineer) having a fair to middling knowledge of electronics, I'm looking for technical info on synthesized speech and musical instrumentation—but only the digital sampling and reconstruction method of real live voice or drums or fiddles or whatever. I know how digital sampling *works*, but who's going to tell me about the actual hardware, such as that used in the Linn drum computer, or Ma Bell's "Sorry the number you have reached...etc"? and, is it possible to get construction techniques on just the sampling circuitry so that I could just hot wire it to a *micro* and make use of its memory to store the data? I enjoyed the digital signal processing series which seems to be similar technology, so c'mon guys, you know what I'm looking for. Who's got the books and the lingo? Gimme some sources.

LARRY NEGRO

db replies:

When a man puts that much faith in you, you just can't let him down. Larry, this must be your lucky month (and magazine). For the whole scoop, see Sidney Silver's article covering electronic devices that speak on page 42*and thanks for the kind words.*

Calendar

MARCH

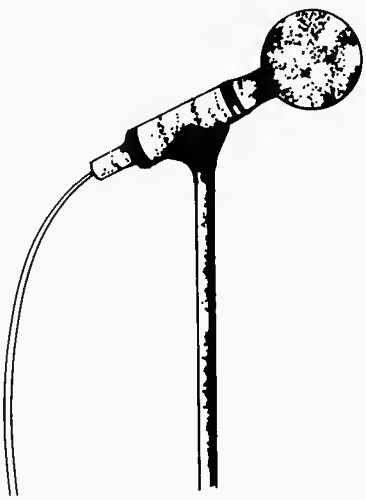
3-6 **Concert Hall Acoustics and TEF Workshop.** In cooperation with V.M.A. Peutz. For more information contact: Don Davis, P.O. Box 669, San Juan Capistrano, CA 92693. Tel.: (714) 496-9599.

14 **LEDE™ Conference.** Eindhoven, Netherlands. Sponsored by Syn-Aud-Con. For more information contact: Don Davis, PO Box 669, San Juan Capistrano, CA 92693. Tel: (714) 496-9599.

15-18 **73rd AES Convention.** Eindhoven, Netherlands. For more information contact: The Audio Engineering Society, 60 East 42nd Street, New York, NY 10165. Tel: (212) 661-8528.

APRIL

23 **MAC 83—Midwest Acoustic Conference.** Hermann Hall, Illinois Institute of Technology, Chicago, IL. The topic will be Audio Signal Processing. For more information contact: Ted Staniec, Knowles Electronics, Inc., 3100 N. Mannheim Rd., Franklin Park, IL 60131. Tel.: (312) 445-3600.



BARRY BLESSER

Digital Audio

The End of Digital Filtering

• It is now time to bring the subject of digital filtering to a conclusion. Using a programmable calculator, we have shown how one can construct a digital filter of the FIR type (Finite-Impulse Response). With a subroutine that generates either an impulse, step or sinewave, we have examined the response with various coefficients. The last filter which we explored had uniform coefficients of 0.1 for all ten taps. This was a very primitive low-pass filter with a very fast transition rate: the ratio of stop-band frequency to pass-band frequency was smallest. Other coefficients in the taps will give larger ratios but lower ripples in the stop band.

It is now time to consider the major parameters in the frequency response of FIR filters in general. We take as the primary categories the concept of pass-band and stop-band. In the idealized world of mathematics, the pass band contains those frequencies which have a gain of 1 (passing the signal); and the stop band contains those frequencies which have a gain of 0 (stopping the signal). In some sense, the 1 and 0 gains are target values. Targets of other than 1 or 0 are less likely but are still relevant. In a special case, one could have a pass band target of 0.5.

For a simple low-pass filter, the pass band region covers the frequency from 0 Hz to some pass band edge frequency f_0; however, for a band pass filter, the pass band might be between f_1 and f_2. With a high-pass filter, the pass band would be between f_0 and f_{max} (f_{max} is the Nyquist frequency). A band-reject filter (analogous to a notch) has two pass band regions, one below the reject region and one above. Such a notch might have a reject region between f_1 and f_2. It would thus have two pass bands—one from 0 to f_1 and another from f_2 to f_{max}.

Clearly, the pass band gain will not equal the target value of 1 but will differ in some way because the actual filter is not a mathematical idealization. The deviation is a measure of pass band quality. This will be discussed shortly.

Like the pass band, the stop band also has a set of frequencies which specify its region. With our simple low-pass filter, the stop band might exist from f_3 to f_{max}. This means that above f_3, the target gain should be 0. A high-pass filter has a stop band from 0 to f_1. We can go on to

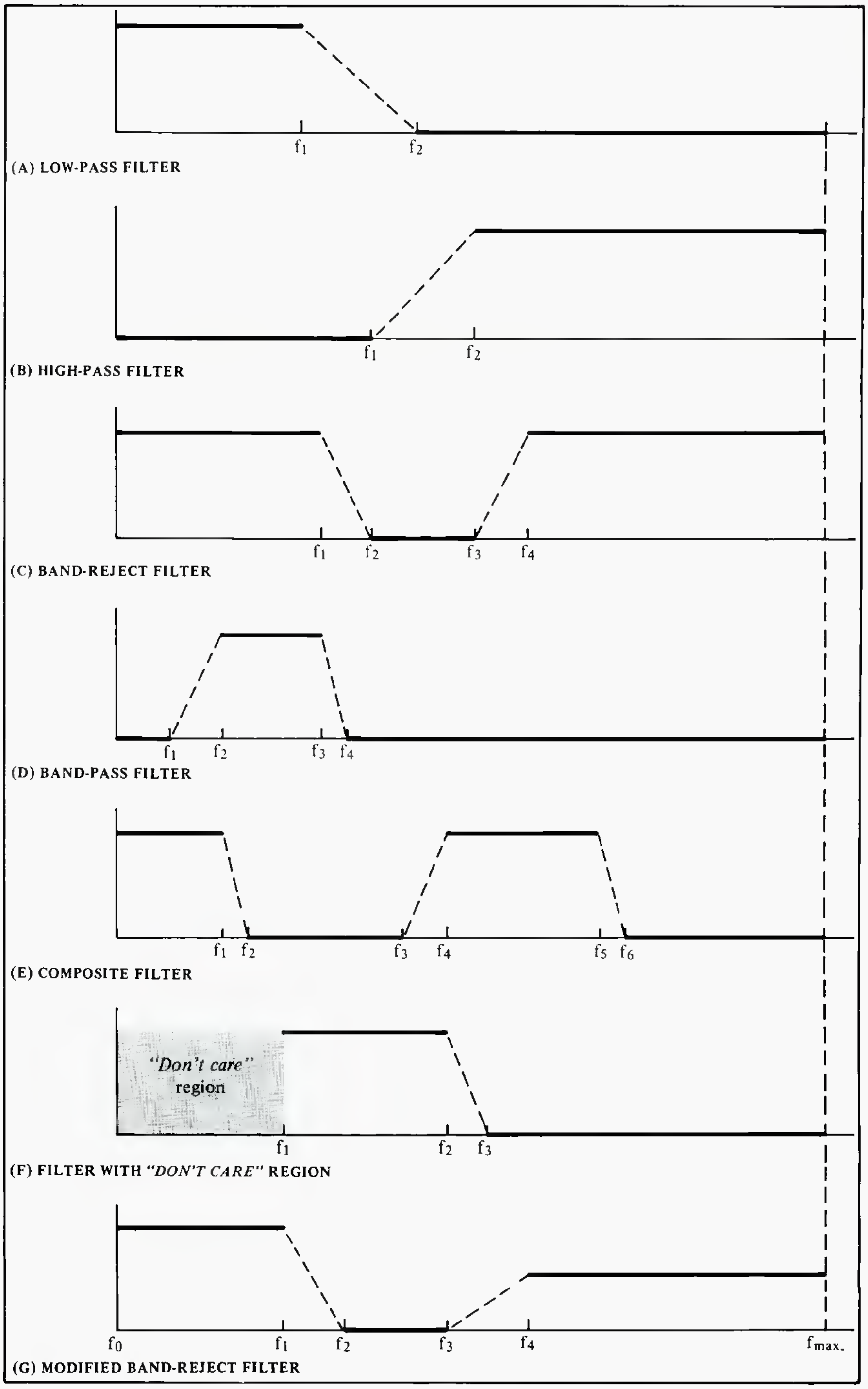


Figure 1. Target values for various types of filters.

define any different set of frequencies for the stop band. With a band pass filter there are two sets of stop bands, one above and one below the pass band region.

In between the pass band and the stop band there must be a transition band. It is unreasonable to have a pass band from 0 to 1 kHz and a stop band from 1 kHz to 10 kHz. How can the filter go from a gain of 1 to a gain of 0 with no change in frequency? Generally, we allow an undefined transition region. Our low-pass filter would thus be stated as a pass band from 0 to 1 kHz and a stop band from 1.5 kHz to 10 kHz. Between 1 and 1.5 kHz, the filter is moving from the idealized gain of 1 to 0. The faster the transition region, the more complex the filter. Very fast filters are just as difficult to build in the digital world as they are in the analog world.

Finally, we come to a new idea in filters: the "don't-care region." In some filter specifications we have no interest in the gain over certain frequencies. We can specify it to be a pass band or a stop band, but the filter will be a better filter if we let the response be an artifact of the remaining part of the filter. This is called "don't care" because we will accept any response.

Examples of the target values for various filters are shown in FIGURE 1. Filter A is the classical low pass with pass band from 0 to f_1, transition region from f_1 to f_2, and stop from f_2 to f_{max}. Filter B is the high pass equivalent of filter A. The stop band is from 0 to f_1 and the pass band is from f_2 to f_{max}. Filter C is the band reject filter in that it rejects those frequencies from f_2 to f_3 and passes those from 0 to f_1 and f_4 to f_{max}. Notice that there are two transition regions which can have different rates.

Filter D is the band pass filter which is the inverse of the band reject. It has two stop bands and one pass band. Filter E has no formal name. It has two pass band regions from 0 to f_1 and from f_4 to f_5 and it has two stop band regions. There are three transition regions.

Filter F also has no name in that it is not a classical form, although it is very similar to Filter A (the low-pass) and filter D (the band pass). The difference is that the region from 0 to f_1 is a "don't care." Such a situation comes from applications where there is no energy in this region because of prior filtering.

Filter G is essentially a band reject except that the two pass band regions have different target values: 1 for 0 to f_1 and 0.5 for f_4 to f_{max}.

These examples only serve to illustrate the kind of freedom that one has in specifying the nature of digital FIR filters.

RIPPLE AND TARGET VALUES

All of the above talked about the target values in terms of an idealization. To what extent will the actual filter realize the targets? For a given number of taps in the FIR delay line, the targets will only be approximate. The word "approximate" is a problem, however. If we wish the approximation to be a good one, we must specify what we mean by it. FIGURE 2 shows various examples of the way in which the approximation may deviate from the target.

FIGURE 2A shows the kind of approximation we get if we demand that the optimization be performed at only one frequency: f_1. Near this frequency, the approximation is almost perfect. All of the derivatives except the highest are 0, and the deviation from ideal is nonexistent until we get far from f_1. This is the so-called Butterworth approximation.

FIGURE 2B treats all of the frequencies in the interval between f_1 and f_2 as being equally important and tries to minimize the peak deviation from the target. This is the equiripple approximation or Chebyshev (in some cases, elliptic). In this example, there are four frequencies which have a maximum deviation from the target, but the deviation is the same for each of these (magnitude).

FIGURE 2C shows the RMS minimization approximation. The "power" in the deviation is minimized. At this point, one

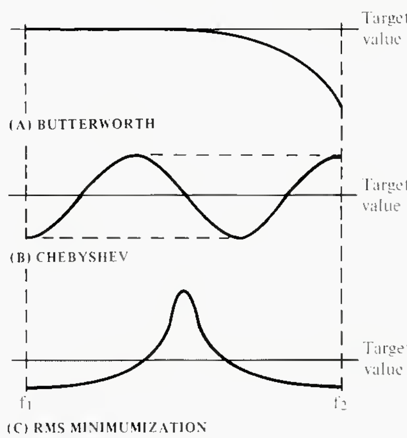


Figure 2. In each of these filter types, the actual response varies from the straight-line target value.

realizes that one can create many other types of approximation measures which can be used in the optimization.

The discussion does not distinguish between targets of 1 for pass band or targets of 0 for stop band. However, the usual notation of dB makes the ripples appear to be different when they are the same. A 1 dB pass band ripple is the same ripple as an 18 dB stop band ripple. When the passband goes from a gain of 1 to 1.12 i.e. a 12 percent change, we have 1 dB. When the same stop band ripple in

The Bose® 802-W.

The smooth solution for your coverage problems.

Conventional horn systems often create more difficulties than they solve. Their sharp-edged radiation patterns make it hard to get sound into all the places you want it—and keep it out of the places you don't want it.

Introducing the Bose 802-W Articulated Array™ Loudspeaker. The full-range reinforcement system with the forgiving dispersion pattern it takes to evenly cover almost any type of seating configuration.

The sloped edges of the 802-W's horizontal radiation pattern will help you prevent the missed edges and slap-echo problems caused by the imperfect fit of horns and rooms.

The gradual "skirts" of its vertical pattern provide exceptionally uniform front-to-back coverage. And the compact, modular design of the 802-W enclosure makes it easy to assemble central clusters that meet the needs of a wide variety of room shapes.

WB-2 Wall Bracket allows 3-axis adjustment.

Send in the coupon for complete technical data on the Bose 802-W Loudspeaker System. Then listen to the 802-W, and hear how Bose's Articulated Array™ system can smooth out the coverage of your sound system installations.

Bose Corporation
Department SE
The Mountain
Framingham, Massachusetts 01701

Please send me complete technical data on the 802-W Professional Loudspeaker.

Name________________
Address________________
City________________
State________ Zip________
Telephone (   )________________

BOSE®
Better sound through research.

linear terms goes from 0 to 0.12, we have a ripple with a peak of −8 dB relative to the pass band. The filters intrinsically use a linear scale even if we report the results in dB.

We must add a footnote at this point. Ripples in either the pass band or stop band appear to be bi-valued, they oscillate around the target values. When they oscillate around the target of 1 it is easy to see, but when they oscillate around the target of 0 they are not easy to see if we plot the frequency response on a magnitude scale. FIGURE 3A shows the ripples as "ringing," whereas FIGURE 3B shows the ripples in the usual magnitude representation. Although the magnitude function removes phase, there is a conceptual advantage in leaving the sign in the function in order to understand the idea of "ringing."

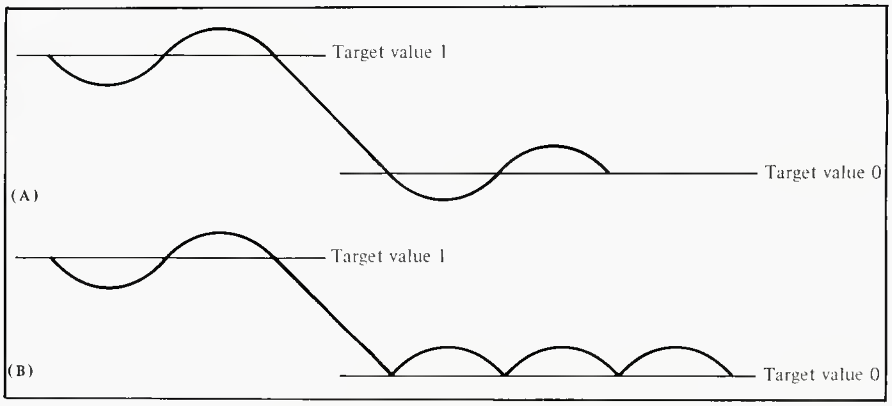


Figure 3. Two examples of ripples at the target values.

It is now time to talk about the interaction between transition rates and "ringing": this is the Gibbs phenomenon.

GIBBS PHENOMENON

The Gibbs phenomenon is a classical set of limits in the design of filters. It says that if we have a very fast transition, then there must be additional ringing or overshoot from the target values. FIGURE 4A shows the increase in the overshoot in going from the first target to the second target as the transition rate is increased. When the transition takes a wide frequency range, the approach to the target is gradual; when the transition is very fast, it overshoots. These overshoots are often viewed as ripples. FIGURE 4B shows the ripples which one gets for the fastest possible transition: there are pre-ripples and post ripples. It is as if the filter has to build up some momentum to get started and then it takes off; when it lands, it takes some time to settle back to the final value.

The classical ripples in the Gibbs effect can be slightly reduced by distributing them over a wider region. The ripple in FIGURE 4B was about 12 percent of the transition amplitude. The highest ripple peaks can be reduced if the lower ripples are increased. This pushes the filter towards the Chebychev approximation.

We are now left with some simple trade-offs. The pass band accuracy and the stop band attenuation both relate to the size of the ripples. The size of the ripples is proportional to the transition rate. Reducing the sharpness of the filter allows us to have better performance in the stop band or pass band or both. Because we can move ripples from the pass band to the stop band with the same transition rate, we can also trade-off stop attenuation for pass band accuracy. These three numbers form the basis for all filter specifications. The number of taps in the filter is the global constant which relates the three parameters. Increasing the number of taps makes a better filter. The extra quality can be used to increase any of the three parameters or a combination of them.

COEFFICIENT IMPLEMENTATION

How do we apply these results to a real filter? The ideas must be translated from these three parameters to an algorithm which creates the coefficients and specifies the length of the filter. Unfortunately, this is highly mathematical and is beyond our scope here. Moreover, for truly optimized results, there is no closed-form solution." A closed-form solution is a mathematical statement that there exists an equation or solvable set of equations which yields a unique result. With a non-closed-form solution, there is only an iterative approximate method requiring a large digital computer which yields the coefficients.

However, because of all the extensive

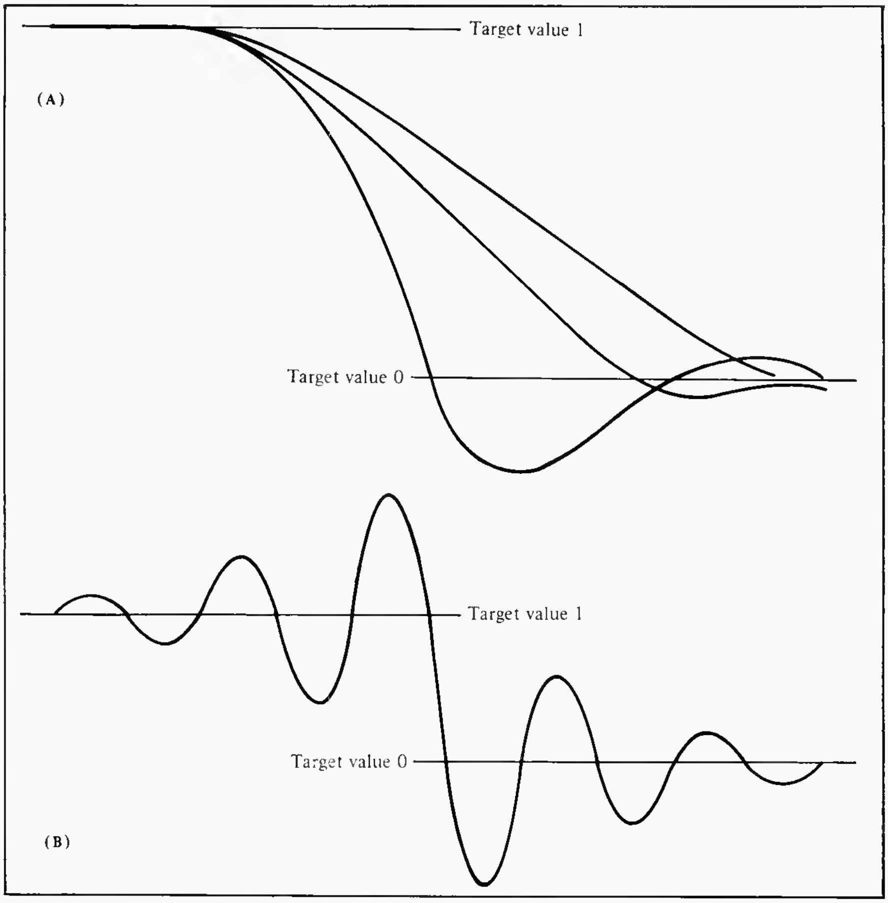


Figure 4. Examples of the Gibbs phenomenon.

work which has been done on digital filters, a computer program, written in Fortran, is available in the public domain and can be obtained from the IEEE in one of their monographs (Programs for Digital Signal Processing, IEEE Press, non-members see John Wiley Publishing Co., 605 3rd Ave., New York, N.Y.). The program is called FIR Linear Phase Filter Design Program by McClellan, Parks and Rabiner. This program is very easy to use. One specifies the number of bands, their target values, the relative weighting of the ripples in each band, and the number of taps in the filter. The program then computes the optimum values of the coefficients and gives the ripples in dB. If the filter is not adequate, one can increase the number of taps and rerun it. Even though it can take several minutes to run a long filter, this kind of "overhead" is irrelevant in a typical development program.

Although the program is intended to run on a main-frame computer, it could run on a personal computer if slightly rewritten. The major issue is the degree of precision in the data words. It is recommended that double precision be used.

This ends our lesson on digital filters for the time being; we may pick up the topic again if the situation appears to be appropriate. ■

KEN POHLMANN

Theory & Practice

The Incompatibility Dilemma

• Back in the old days life was pretty simple. On Sunday afternoon you got into your Chevy and went for a drive with all the other Sunday drivers. If you wanted to listen to some music, you turned on the radio and the sound coming out of the speaker in the dashboard sounded great. Maybe you stopped for gas—the attendant turned on the pump, put in a few gallons, checked the oil and cleaned the windshield. Then you paid him a buck and drove off. When you got back home, you turned on the television, but all three channels were uninteresting. So you put a record on the record player and turned the volume low and sat by the cabinet so you wouldn't disturb the rest of the family. Then you dozed off. Whe the stylus came to the end of the record, it went cha-chick, cha-chick, cha-chick....

Today is Sunday, but you are up early anyway: the kids put a metal frying pan in the microwave oven—it was incompatible and the oven fried itself with a bang. Then your son asked for a lift to the computer store, explaining that his motorcycle has a loose brake caliper and he needs a metric allen wrench to fix it—all of yours are incompatible English sizes. You all piled into your Mazda and took off. At the computer store your son bought his software and your young daughter got a video game; they wouldn't take Diner's Club, so you borrowed your son's Visa. On the way back you stopped for gas, but the station was out of unleaded premium gas offering only incompatible unleaded regular, leaded regular, and gasohol. You stopped at another station and when you got out to pump it, you noticed that your gas cap was missing. The station owner tried over two dozen varieties, but they were all incompatible. You filled up anyway, but your gas credit card was incompatible with that station, so you forked over $20 in cash and had to borrow $1 from your daughter (she made you sign an IOU). You stuffed a rag in the filler neck and drove off looking like a big Molotov cocktail. Back home, you settled down to watch a good movie on TV, but you had misread the listing—it was on a scrambled channel and you lacked a decoder to make it compatible. You tried the video tape you bought last night, but soon realized it was incompatible Beta instead of VHS. Then your daughter came in, crying: she had mistakenly bought an incompatible Colecovision cartridge instead of an Atari. Then your son called you into his room: the new software wouldn't run. A few fun hours later, your wife pointed out that the package said 6502 whereas the computer, she believed, was Z-80. Then a friend of your son's showed up with a new graphics card he wanted to try. When it was inserted there were some leftover fingers on the socket—the card must not have been compatible because when you powered-up, the computer burst into flames. You grabbed a fire extinguisher, but it clearly said do not use for electrical fires—it was clearly incompatible. You hustled your family outside and a fire engine arrived, but the developer had put incompatible fire hydrants in the subdivision—the hoses wouldn't fit. You watched as your new house burned down and wondered: Is it worse to play a Dolby B cassette without Dolby to make it sound brighter, or to play a non-Dolby cassette with the Dolby in to cut the hiss?

An early incompatible design

In my opinion, much of the charm of modern living lies in its incongruence. Frankly I think it's great to be almost totally out of touch with everything that's going on; it's better that way. I mean, seemingly out of necessity our lives are filled with inconsistent and incompatible things that we calmly take for granted; we don't try to explain them, or to understand them. That's the essence of an advanced technological society. Can you explain the operation of the International Monetary Fund, or how offset printing works? Or in your own field—can you draw a schematic for the circuitry inside the op amps in your console; can you write the firmware to program a microprocessor-controlled autolocator; can you explain why a JFET amplifier sounds different than a bipolar amplifier? Quick, what's the horizontal sync frequency in the NTSC standard?

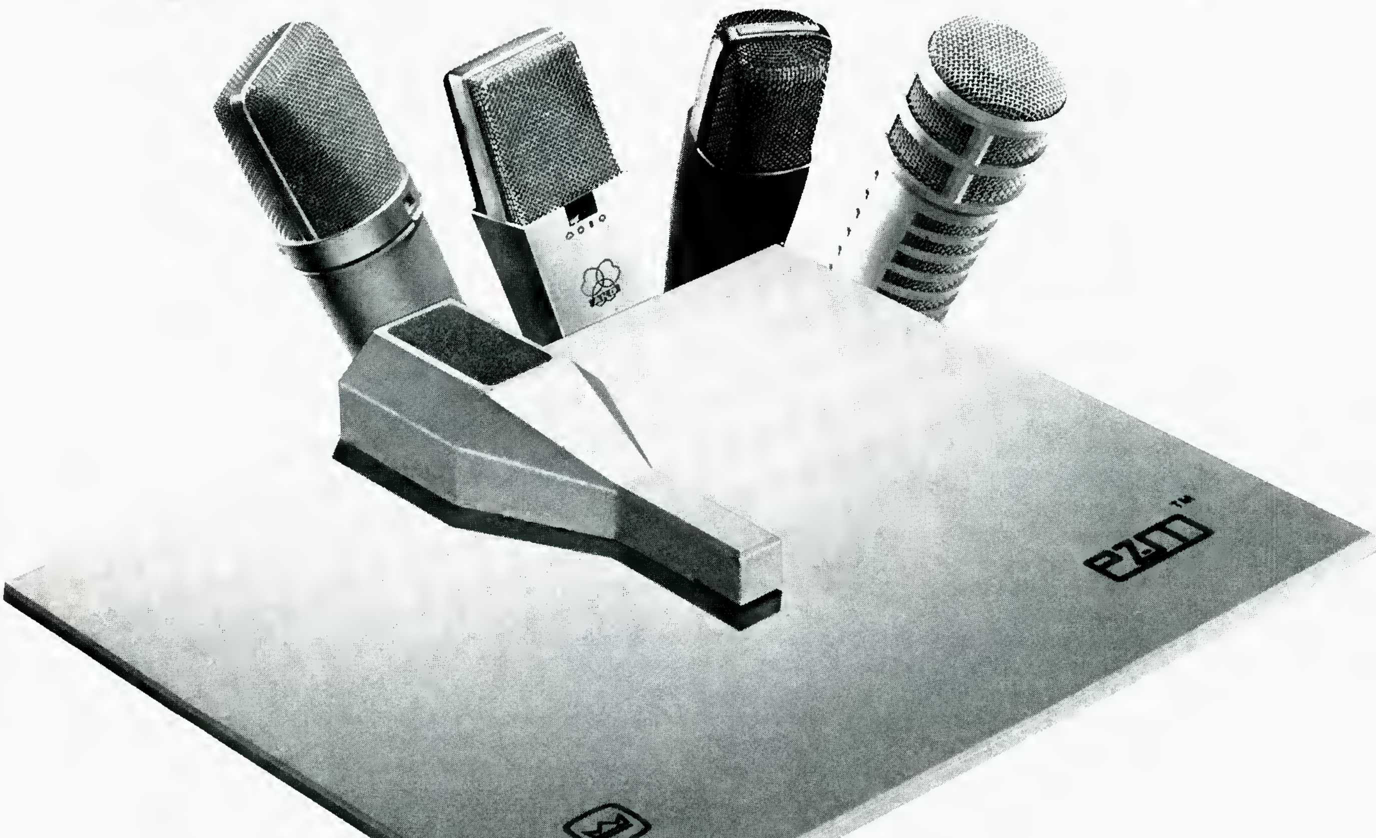
Few of us could correctly respond to all of those questions. While we could learn the answers if we had to, it's obviously silly to encumber ourselves with all kinds of extraneous understanding. Modern life doesn't demand it, and in fact, even encourages us not to try. You see, as our environment becomes more and more sophisticated, the proportion of the environment we understand rapidly decreases. Ultimately, we will all be experts on miniscule, mutually-exclusive subjects.

I've digressed long enough—what is the point? The point is that we all come to accept incredible inconsistencies in our perception of things. We can't understand things now, so why not welcome anything else that's new and even less understandable? The fact is that anything that is new and improved and more difficult to understand is automatically labeled as being better. And that sets us up for the big dilemma: We place tremendous importance on things new and different, and the best way to make something be perceived as new and different is to make it incompatible. That ingrained perception both devalues anything that already exists and guarantees its obsolescence. I won't venture to say that we produce incompatibility for incompatibility's sake, but I will state that we value incompatibility, and have evolved a system which self-obsolescence nourishes and evolves itself on its own incompatibility.

That tyranny is everywhere and is so predominant that we routinely fail to even notice it. Consider—Alpha and Omega make picture frames. Omega introduces pretty pictures which don't fit in Alpha's frames—they only fit in Omega's frames. Everyone buys Omega pictures and frames. Alpha introduces even prettier pictures, but they don't fit Omega's frames or Alpha's old frames—they only fit Alpha's new and improved frames. Everyone buys Alpha's new pictures and frames. The familiar process continues, back and forth. Like it or not, we must conclude that it is cruelly-inflicted incompatibility that propels the system onward.

Curiously, the system is a success, and is apparently good for everyone concerned (the pictures get prettier and

STAR PERFORMERS



Crown PZM® microphones are found in good company. Recording studios and sound engineers concerned about quality include PZM in their inventory of "best" mikes.

There are good reasons for that.

The extraordinary reach and clarity of PZM mikes adds extra dimensions to miking set-ups, and can often simplify the set up. An overhead PZM, plus a ribbon or condenser mike for extra emphasis on the snare, is often all you will need for a drum kit.

PZM can also add quality to the recording. One skeptical symphony orchestra conductor listened to a tape recorded with only two overhead PZM mikes, and joyfully admitted that it was the first time anyone had recorded what he heard on the podium.

The low-profile, "hidden" look of PZM mikes makes them ideal for podiums, especially on TV.

PZM is also becoming the microphone of choice for theatrical productions, especially musicals. 180° pickup with no "off-axis" problems, and accurate pickup at over 30 feet, make them indeed worthy of top billing.

Send for your free copy of the latest issue of PZMemo, the Crown newsletter of PZM ideas. Among other worthwhile items on how and where to use PZM, you'll soon be reading the details of a new, totally redundant clip-on PZM mike. To stay up-to-date with PZM technology, just fill in the blanks below and send this corner of the ad to Crown. It could get you star rating on *your* next performance.

Name ______________________________

Address ____________________________

City ________________ State ______ Zip __________

crown®

...WHEN YOU'RE READY FOR REAL!

1718 W. Mishawaka Rd., Elkhart, IN 46517, (219) 294-5571

Circle No. 21 on Reader Service Card

prettier), at a price. The consumer pays; the company profits and builds better (and incompatible) products; the consumer pays again, for a better product. The bottom line is that new technology pushes aside the old. Most of us agree that this is good. We have to embrace newness, proliferation, and incompatibility. Are we agreed? Is it worth it? Let's test the Incompatibility Theory.

THE INCOMPATIBILITY THEORY

The new Compact Disc is the most incompatible thing I can think of. As far as I know, there isn't a single typical music lover in the world who can listen to a Compact Disc. It doesn't make any difference how much he paid for his stereo, or how new it is — he can't play a Compact Disc. If he wants to play a Compact Disc, he will have to go out and buy a new Compact Disc player. And after he buys a new Compact Disc player, he won't be able to play any of his old records on it — he'll have to buy all new Compact Discs. And something else: Not only is the Compact Disc incompatible with all old technology, it's also incompatible with all *new* technology. That's right. Every recording studio owner who has gone out on a limb with a 16-bit, 48 kHz sampling rate digital machine has a big problem: the Compact Disc is 44.1 kHz. That's it, that's the price of incompatibility. Is the price fair?

The price is fair, and the incompatibility is worth its trouble, if the product warrants it. And the consumer is a fiend in these evaluations. Only a fraction of new products succeed—the consumer is ruthless when it comes to this kind of decision-making. So what about this Compact Disc? Will the consumer embrace this greatest of all incompatibilities—the incompatibility which will obsolete millions of dollars of record players and multiples of millions of dollars of record collections—will the consumer buy such a player and begin his music collection again from scratch, often buying identical recordings over again, on the new format? Will the professional recording industry adapt a dual standard of 48 and 44.1 kHz, a duality which will demand a legacy of conversion, and possible obedience to a Sony edict that all Compact Disc masters adhere to a ¾-inch U-matic video protocol? The consumer will indeed do this, and the industry will indeed comply, if the Compact Disc deserves it.

So, just how good is it? After a century of universally unsatisfactory disc media, the Compact Disc system offers something better. It is a digital system, using a 44.1 kHz, 16-bit linear quantization per channel. The modulation system employed is a newly-developed EFM (Eight-to-Fourteen Modulation) which

Still incompatible after all these years permits a high data density of 43,218 bits per second. Its digital advantages are apparent: a flat frequency response from 20 Hz to 20 kHz that is a lot flatter than any analog flat response ever was; a signal-to-noise ratio greater than 90 dB; dynamic range greater than 90 dB, channel separation greater than 90 dB. Harmonic distortion is less than 0.05 percent at peak modulation and wow and flutter is essentially immeasurable. There is no rumble. Playing time is a maximum of 80 minutes per side, two- sided discs being available. A four-channel disc would have half the playing time. The 4.7-inch diameter disc is faced with a protective transparent coating to prevent all but the most severe damage to the disc from ever being audible.

The pickup consists of laser light, operating at a wavelength of 780 nanometers with an ALGaAs laser with a focal length of 2 micrometers. That's light reading the data—not a diamond (world's hardest substance) scraping along a vinyl (world's softest substance) groove. Speaking of grooves, there aren't any on the Compact Disc. Instead there is a helical track of pits carved into the flat disk, each pit about 0.6 microns wide and 0.2 microns deep. A typical disk might contain over six billion such pits, each representing one binary bit. The laser scans the track at approximately 4.3 million bits per second. Additional bits will permit programming of titles and texts for video display, as well as user-defined programming of selection playing sequences and repeats.

The Compact Disc, if not physically abused, will never wear out or show signs of deterioration. To guard against abuse, as well as unavoidable dust and dirt, a sophisticated error correction system is incorporated in the player. The error-correction scheme uses a CIRC Cross-Interleave Reed-Solomon Code employing delayed interleave and two sets of delayed parity words which correct errors as large as 3,500 bits, and linear interpolation which can audibly conceal errors as large as 14,000 bits. Holes with a diameter of 2 millimeters could be drilled through the disc without incurring audible losses. In addition to correcting errors due to disc abuse, the CIRC scheme will minimize any manufacturing defects. As analog records have long reminded us, no mass-produced item is always free of defects, but with the Compact Disc most such defects will heal themselves through correction algorithms during playback.

The Compact Disc is envisioned as the standard home playback medium, as well as in applications such as automobile sound systems and portable Walkman-type units. Since its introduction in 1979 by Philips, and joint development with Sony, the Compact Disc has been endorsed by manufacturers such as Bang and Olufsen, Crown, Dual, Matsushita, Nakamichi, Onkyo, and Studer/Revox. In addition, the hardware is supported by the software conglomerates of Polygram (Philips, Phonogram, Polydor, London Decca, Deutsche Grammophon), CBS Sony, Warner/Pioneer, Nippon Columbia, and others. Price for a Compact Disc will be competitive with a conventional analog disc, and the player will be priced in the middle hi-fi market.

That's the Compact Disc. Is it worth it? You decide. If you've been laid-off from Chrysler for two years, you're sure as hell not going to run out and buy a Compact Disc just because it sounds a little better than your old record player. But just as the LP displaced the 78 RPM, the Compact Disc will probably displace the LP. It is superior to any other system, and deserves to be established as the standard for the future. And it is incompatible as hell. I guess that's the ultimate irony, and comfort. When something is clearly superior, it has no choice but to be incompatible. It's the same thing when you're running a race—the winner is incompatible with all of the losers.

So, is incompatibility the burden we must bear to enjoy the pleasures of advancing technology? The answer is a definite yes and no. While innovation apparently yields incompatiblity, there's nothing to prevent other innovations from yielding...compatibility. But that's next month's column. ■

LEN FELDMAN

Sound With Images

Colored Sound

• At the recently concluded conference of the Society of Motion Picture and Television Engineers (SMPTE) Grumman Aerospace demonstrated a unique new method for broadcasting stereo audio for television. The system is called Rainbow Sound, and the reason that name was chosen will soon become obvious.

The Grumman system is simple and inexpensive and is made possible because Grumman's SYNC PROC™, a product which eliminates shift problems in match-frame editing, has now been combined with a color encoder for graphics generation synchronization. This equipment enables Grumman to encode the second channel of audio into a *video* format and switch it into the video signal.

I'll explain the operation of the system in more detail, but before we get into that I should point out that there are some non-technical problems related to this system which may well prevent its use or acceptance by the TV broadcast industry. As most of you know, if you've been following this column or reading the many articles concerning the ultimate coming of stereo audio for TV, the EIA (Electronic Industry Association) Multi-Channel TV Subcommittee is nearing the end of its four-year effort to recommend a standard system of stereo audio for TV. While the FCC may not endorse the system chosen by industry because of their new "let-the-marketplace-decide" policy, if the EIA committee can agree upon a single system, it is likely that the system chosen would become a de facto standard—especially if all major segments of the industry endorse it in advance. With that in mind, it seems very late in the game for another proponent to enter the fray, and my personal opinion is that Grumman has little chance of getting approval for their system unless they move very quickly. Still, the system is an extremely interesting one, so let's take a look at how it works.

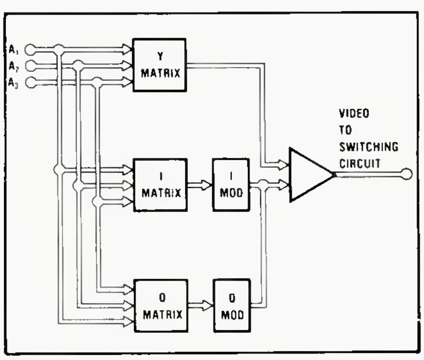


Figure 1: Samples of the stereo difference signal (L-R), shown here as A1, A2 and A3, are processed much as R,G,B color signals would be sampled by the Grumman Color Encoder.

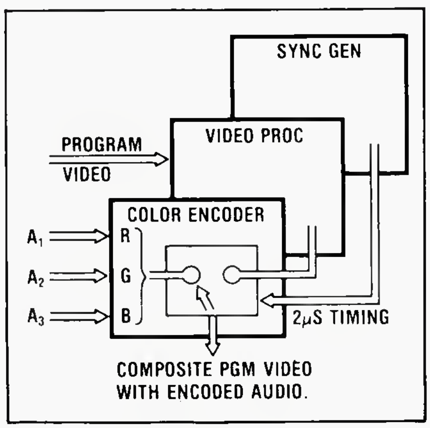


Figure 2. The encoded, time-compressed audio signal is joined together with the composite video program, becoming part of the video format.

COLOR ENCODER DOUBLES AS AUDIO SAMPLER

Some time ago Grumman developed a color encoder to support video graphics generation with all timing requirements met, and with NTSC encoding of RGB (Red, Green, Blue) graphics output. The system also provides Y (luminance signal), I and Q (chrominance or color signal) outputs from the color encoder. It is this very same system hardware that Grumman has adapted for their stereo audio system.

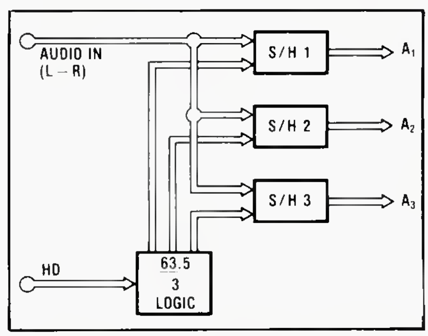


Figure 3. Sample & Hold (S/H) circuits sample the amplitude of the stereo difference (L-R) signal three times during each horizontal line of video.

In essence, the audio signal is sampled and the samples are routed to the RGB inputs of a standard color encoder where they are matrixed into a standard NTSC composite form, as illustrated in FIGURE 1. The encoded color frame that results from this process represents a stereo difference signal (L – R). Program video

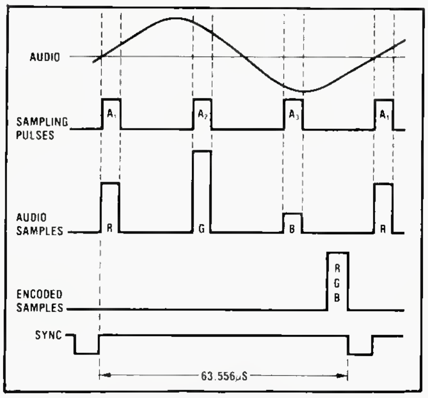


Figure 4. Audio waveform (top) is sampled three times per horizontal video line (bottom) and encoded into a single, 2 usec signal, identified here as "RGB."

and the "color-encoded" (L – R) audio are joined at the end of each horizontal line of video. This action takes place in the color encoder (FIGURE 2) by switching away from program video to the color-encoded audio for 2 microseconds prior to the leading edge of the horizontal synchronization pulse.

Once the L-R signal has been created, that signal is routed from the audio encoder to three sample-and-hold circuits, as illustrated in FIGURE 3. The time rate for one horizontal line of video within the NTSC format turns out to be 63.5 microseconds. (There are 15,734.264

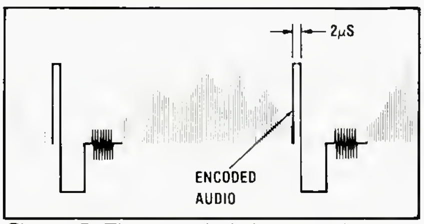


Figure 5. The encoded (L-R) audio is inserted into the composite video signal at the end of each line, as shown. It reduces useful picture width by a negligible amount, since most TV sets over-scan anyway.

horizontal lines per second in the standard NTSC color TV system used in the U.S., and the reciprocal of that number works out to be 63.5 microseconds per horizontal line.) If three samples of audio amplitude are taken during the time span of one horizontal line, then we have a sample approximately every 21 microseconds, as shown in FIGURE 4. That works out to be a sampling rate of approximately 47.2 kHz, or more than enough bandwidth to handle audio frequencies up to 15 kHz, the present legal upper limit for audio on FM and TV. The three samples per line are encoded into the same sort of RGB format that Grumman's color encoder would normally create for separate R, G, and B color signal inputs and compressed into a total time span of only 2 microseconds, as illustrated by the line labeled "Encoded Samples" in FIGURE 4.

The 2-microsecond encoded audio sample is now switched into the program video in the 2 microseconds just preceding the leading edge of the horizontal sync pulse, as shown in FIGURE 5. In effect, the stereo audio (L-R) information actually uses up a very small bit of picture time, but it is located at the very end of each horizontal line—an area that is almost always off-screen because TV sets are designed with a certain amount of overscan in the first place. Incidentally, in case you are wondering where the name "Rainbow Sound" comes from, it was given that name because in implementing the system, a narrow color stripe appears at the extreme right of the video picture. It's not visible, of course, unless the horizontal deflection system of your TV set is unable to sweep fully across the screen.

In their description of the system, Grumman is quick to point out that the value of the audio signal derived from the 2 microsecond sample will only be correct if the settings of luminance gain, chrominance level, and chrominance phase are correct. The system should not be confused with any sort of PCM approach, where signal amplitude errors are of little consequence and variations in pulse *width* convey information. The Grumman approach might best be described as more of a PAM (Pulse Amplitude Modulation) scheme than as a PCM system.

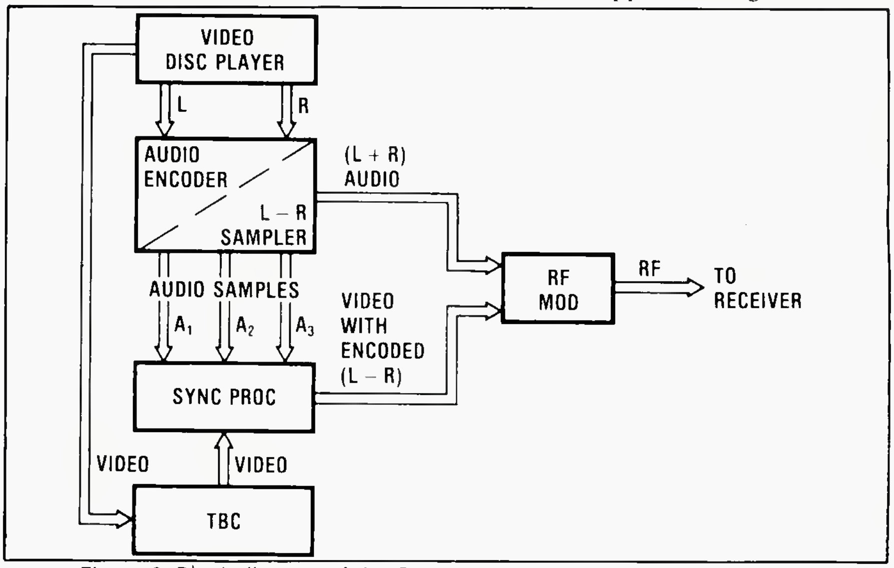


Figure 6. Block diagram of the Grumman stereo audio encoding process.

A detailed representation of the encoding process is shown in FIGURE 6. Using a videodisc as a video-and-stereo-audio program source, the video is processed through a time base corrector (TBC). Left and right stereo signals are converted into sum-and-difference signals. The L + R signal joins the video signal as the primary modulation of the transmitter. The stereo difference signal (L – R) is processed as described earlier. The program video with the encoded L – R, along with the L + R audio channel, are routed to an RF modulator and then to the antenna terminals of the TV receiver in this closed-circuit demonstration.

DECODING THE SIGNAL

Some modification of a typical home

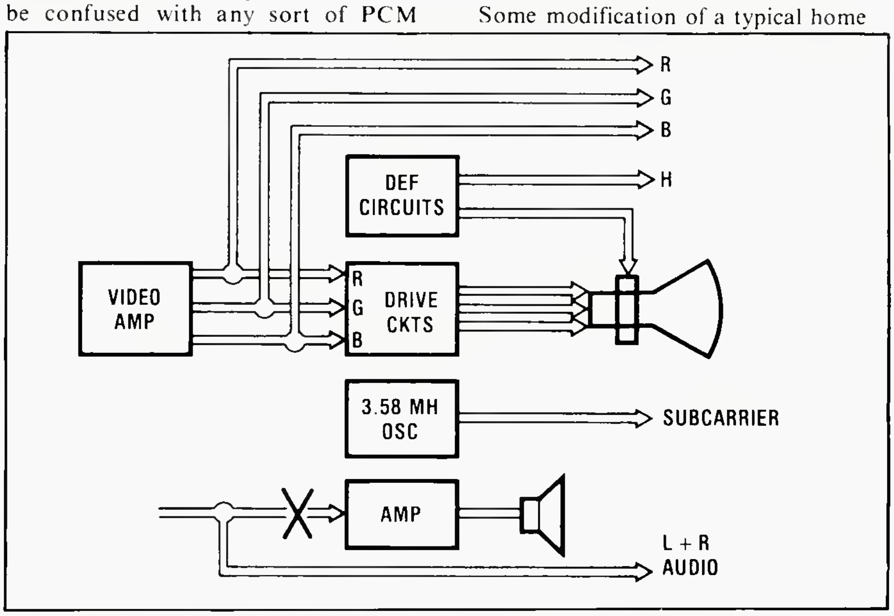


Figure 7. R,G,B Horizontal and Subcarrier take-off points would have to be accessed in existing receivers to work with the Grumman Rainbow Sound system. Available (L+R) audio would have to bypass TV set's audio system and be fed to external matrix decoder.

TV receiver would be needed in order to decode the stereo signal transmitted in this way. This modification would consist primarily of tapping into the horizontal sync signal, the color subcarrier, and the separate Red, Green and Blue feeds, as illustrated in FIGURE 7. The monophonic audio channel L + R is broken and brought out of the set for subsequent matrixing with the recovered L – R signal. The three audio samples recovered from the TV receiver as individual R, G and B signals are routed to sample-and-hold circuits. Timing logic derived from the horizontal sync pulse and the color subcarrier provides the necessary timing to restructure the three samples (per line) 21 microseconds apart. The signal is then filtered by means of a low-pass filter which recreates a continuous audio waveform which is then amplified as the recovered L – R audio channel. This part of the process is illustrated in FIGURE 8.

Once the L – R signal has been recovered, it is matrixed with the already available L + R signal in the familiar manner currently used in broadcasting stereo FM. That is, the L – R signal is added to the L + R signal to recover the left channel ([L + R] + [L – R] = 2L), and the L – R signal is subtracted from the L + R signal to recover the right channel ([L + R] – [L – R] = 2R).

OTHER PROBLEMS OVER THE RAINBOW

Aside from the problem already discussed (its lateness in arriving upon the scene), there is another problem which Grumman would have to solve before serious consideration could be given to their Rainbow Sound system. While the Rainbow Sound system could certainly be used either for transmitting stereo or for transmitting a second language, it seems clear from the description offered by Grumman that both of these services could not be implemented simultaneously by a single station. Yet all of the systems now under consideration for multi-channel TV audio are able to provide stereo service *and* a separate audio program (bilingual, or even a completely unrelated audio service) at the same time. That ability is, in fact, listed as one of the necessary criteria for any system being considered by the industry committee studying the other three systems.

In short, Grumman's late entry into the stereo audio for TV competition may have come too late to do them, or the industry, much good. ■

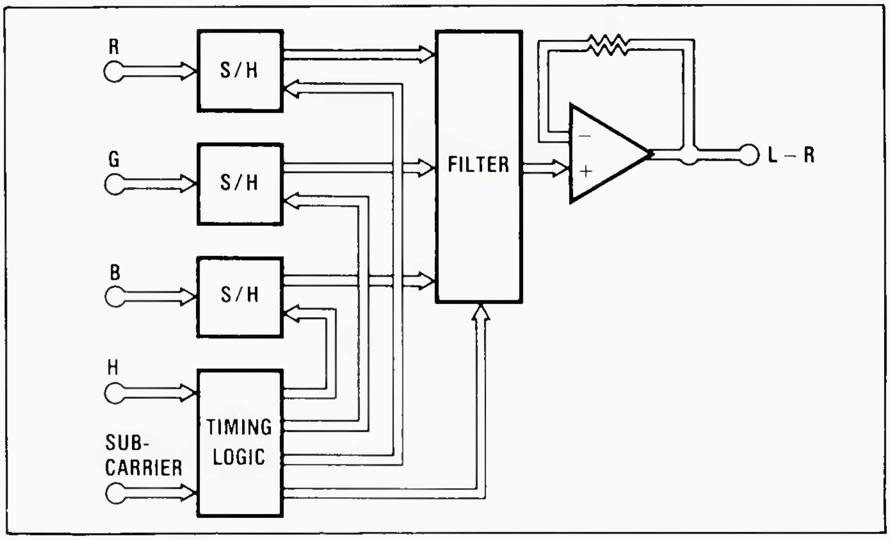


Figure 8. Elements of the decoder required to recover the (L-R) stereo difference signal before re-matrixing with the available (L+R) signal.

JOHN EARGLE

Sound Reinforcement

Extending Power Bandwidth

• We have all seen power amplifier specifications which show frequency response at low outputs extending from perhaps a few hertz up to 100 kHz or beyond. When the same amplifier is driven to full output, that flat frequency response will likely be quite limited, especially at the high-frequency end, due to the inability of the output devices to switch large amounts of current quickly. In making this observation, we state that the power bandwidth of the amplifier is limited at high output levels.

While not exactly analogous, the response of a good cassette recorder, at levels 20 dB below full output, will likely be fairly flat out to 20 kHz; at full output, the high-frequency response may start to roll off below 10 kHz. In this case, the problem is tape saturation—the inability of the tape to accommodate the full-level signal boosted at high frequencies in the record amplifier.

Power bandwidth considerations are important in high-level sound reinforcement, due both to voice coil heating problems as well as cone or diaphragm displacement problems. This will often lead to division of the spectrum into three or more bands, as may be required by the program at hand.

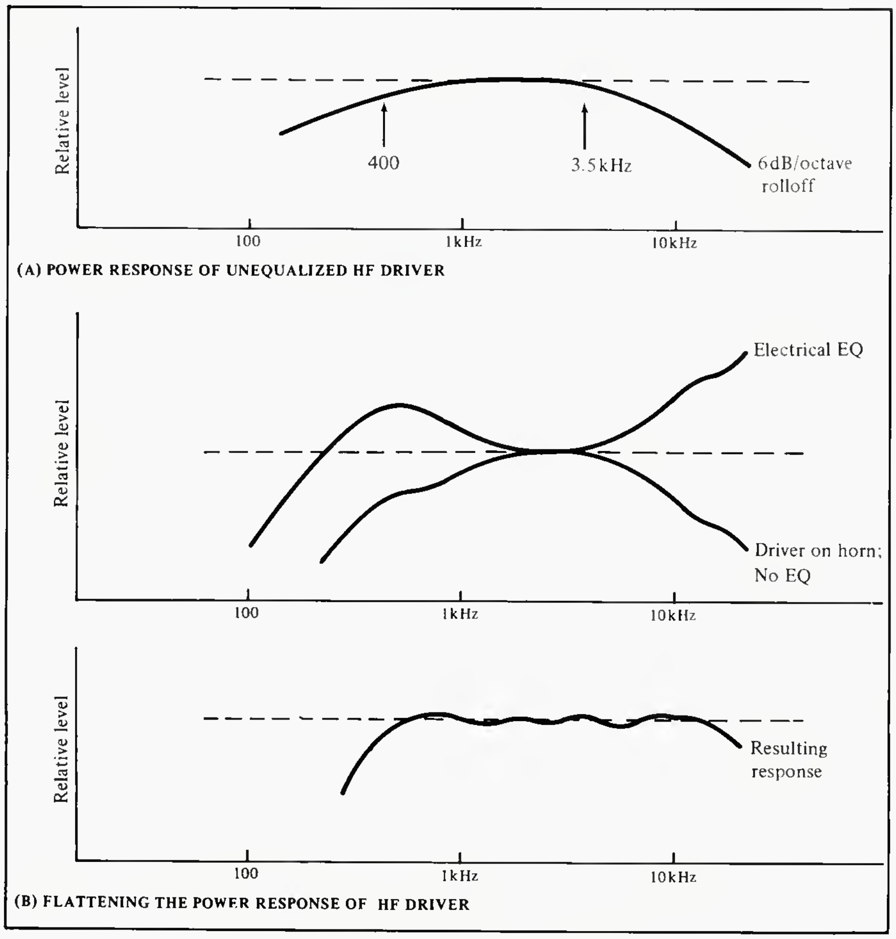


Figure 1. High-frequency Power Response.

THERMAL AND DISPLACEMENT OVERLOAD IN HIGH-FREQUENCY COMPRESSION DRIVERS

FIGURE 1A shows the unequalized power response, as it would be measured

on a plane wave tube, for a typical high-frequency compression driver. In a normal application, as shown in 1B, the driver is placed on a constant coverage-type horn, electrically rolled off at the low end and boosted at the high end, giving net flat response both on- and off-axis.

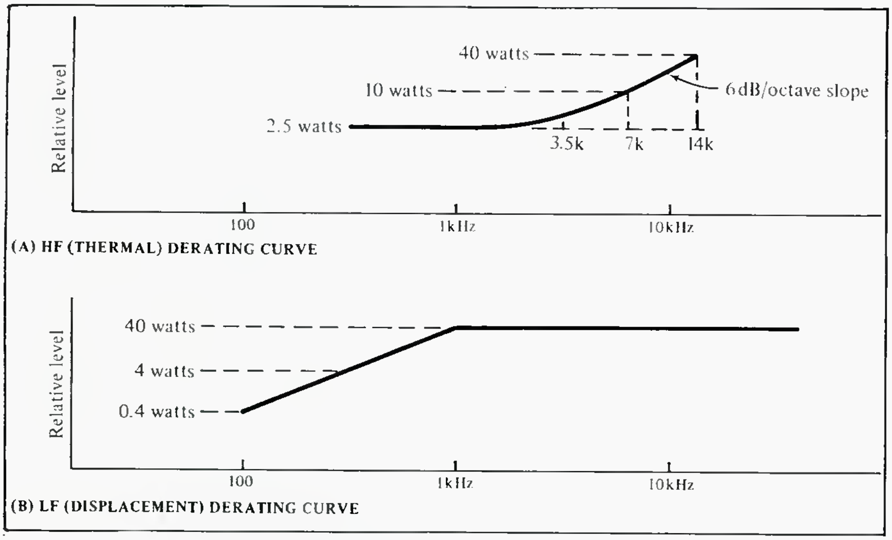


Figure 2. High-frequency de-rating curves.

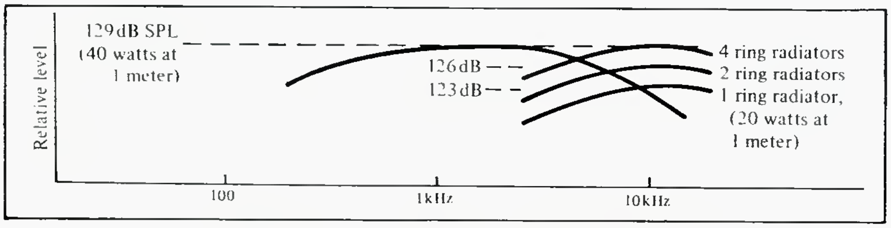


Figure 3. Implementation of a UHF system.

As long as we operate the horn-driver combination at "safe" levels, then all will be well; we will be able to get flat response out to 12-15 kHz while maintaining a crossover frequency in the 500-800 Hz range. Let us assume that the driver has a thermal power rating of 40 watts. What this means is that the voice coil can safely handle input power as high as 40 watts—provided that there are no diaphragm displacement problems that could result in shattering of the diaphragm.

Now let us see what happens as we begin to increase the power to the combination. If we put in 40 watts at, say, 1.5 kHz, the driver can handle it in stride. However, if we raise the frequency of our program input to 5 kHz, then the high-frequency boost will increase the signal fed to the driver up to about 80 watts. While the driver may be able to accommodate short, momentary bursts of 80-watts input, it certainly cannot sustain a continuous input at that power.

Depending upon how high in frequency we wish to maintain flat response from the driver, we must apply some kind of de-rating curve to it as far as mid-band power is concerned. For example, if we want to maintain flat response out to about 14 kHz, then we must apply the de-rating curve shown at FIGURE 2A. What this tells us is that we can put full power into the driver at 14 kHz only if we are willing to restrict mid-band continuous power input to 2.5 watts or less! Conversely, if we wish to maintain full mid-band power input to the horn driver combination, we must be careful not to boost program content above 3.5 kHz, in effect limiting the HF power bandwidth of the driver.

If we choose to do this we may run into another problem, that of displacement overload at frequencies below 1 kHz. FIGURE 2B shows a typical displacement de-rating curve for a 10-cm (4-in.) aluminum diaphragm driver. This curve tells us that if we wish to enjoy input powers as high as 40 watts at mid and high frequencies, then we must cross the driver over in the 1 kHz region if we are to avoid diaphragm fracture due to excessive excursion. We are, in effect, limiting its low-frequency power bandwidth.

For many indoor applications such as motion picture theater work, we may safely cross the driver over at 500 Hz as well as equalize the horn driver combination flat out to 14 kHz without encountering the problems we have been discussing here. It is in high-level music reinforcement applications where we really begin to encounter power bandwidth problems with high-frequency drivers.

THREE- AND FOUR-WAY SYSTEMS FOR MUSIC REINFORCEMENT

The problems we have discussed dictate the use of additional spectrum division. What we need is another set of transducers capable of assuming the load above 3 kHz, as well as some kind of

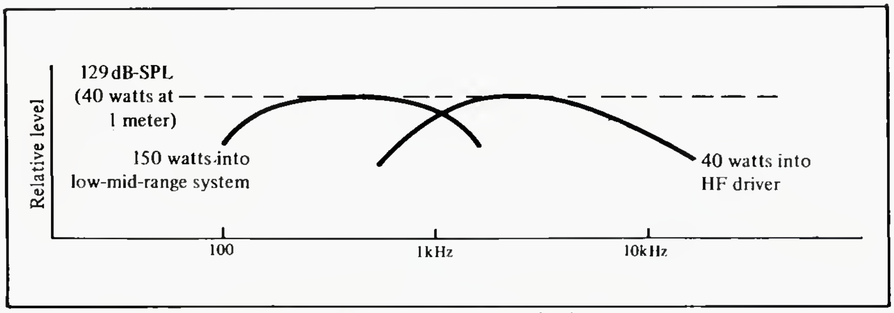


Figure 4. Implementation of a lower mid-range system.

upper-bass or lower-mid-range system to cover the range from 200 or 300 Hz up to 1500-2000 Hz.

Ultra-high-frequency (UHF) transducers, usually in the form of ring radiators, are available from a number of manufacturers. Typically, a single ring radiator can handle continous power inputs of 20 watts above 3 kHz, and their sensitivities, depending on their particular radiation patterns, are usually in the 105-110 dB range, 1 watt at 1 meter. They are most often used in multiples. A good rule of thumb is that there should be four of them for each high-frequency horn/driver combination in order to maintain flat power response out to 15 kHz, while at the same time maintaining full mid-band input to the high-frequency driver. FIGURE 3 shows the implementation of four UHF radiators with a single high-frequency horn driver combination.

The coverage of the upper-bass region

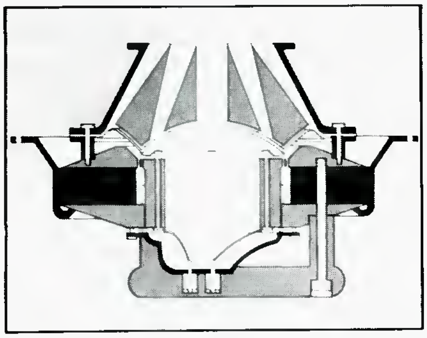


Figure 5. The Community Light & Sound M4 mid-range compression loudspeaker.

requires componentry which has only been available in recent years. Many sound contractors who are in the music reinforcement business have designed their own small front-loading horns for high-sensitivity 30-cm (12-in.) transducers originally intended for musical instrument (MI) use. In their normal application, these transducers exhibit sensitivities of 103 dB, 1 watt at 1 meter, and can handle power inputs of 150 watts. When mounted in the horn enclosures, the axial sensitivity typically increases to about 107 dB, 1 watt at 1 meter. The increase in sensitivity is due both to horn loading and an increase in directivity index. One such unit is a good match for a typical 90-by-40 degree high-frequency horn/driver combination, since the difference in power handling just about compensates for the sensitivity difference (113 dB, 1 watt at 1 meter for the high-frequency system). FIGURE 4 shows the implementation of such a system.

Along different lines, the Community Light and Sound Company has developed a special upper-bass system which has a sensitivity of about 112 dB, 1 watt at 1 meter, and a continuous program power rating of 200 watts. Their system, the M4, uses a 7-in. diameter diaphragm feeding through a conventional annular slit phasing plug into a 4-in. throat coupled to a horn designed to go down to 200 Hz.

Depending on the musical requirements, low-frequency systems may need to be supplemented below their normal 40 Hz limit through the use of sub-woofers. We will cover that application of power bandwidth extension in our next column. ■

Editorial

Crime and (No) Punishment

THE TIMER ON our VCR is acting up again. We had wanted to record a National Geographic special while we were downstairs making dubs of some master tapes to sell to a friend, but the damn thing didn't come on at the right time. We missed the special and were left with nothing more than the six o'clock news. And depressing news at that: some kids had figured out how to make their own microwave antennas and were cheating the local HBO out of its rental fees.

We hit the eject button, and turned on the home computer. We had just borrowed some software from Harry Hacker down the block, who said we could copy anything we liked. Maybe playing with a new graphics game would help take our minds off of all this dishonesty.

Then our associate editor called and ruined the whole evening by reminding us that the editorial deadline had long since come and gone. He suggested turning off the color graphics and turning on the printer. Not a bad idea at that—we had just swapped a few master tapes for a new word-processing disk, and this would give us the chance to try it out.

We had recently received a copy of the minutes of the last AES Digital Audio Technical Committee meeting, which was held just prior to the 72nd AES convention in Anaheim (October 22, 1982). Since this issue of **db** will be distributed at AES 73 in Eindhoven, maybe we could find something relevant to comment on there.

Guess what!? The report stated that incoming AES president Dr. Tom Stockham had suggested the Society might give some attention to the problems of commercial counterfeiting of recordings, home tape copying, and the impact of digital technology on it all.

During the committee meeting, it was noted that most industry attempts at various copy-prevention plans had failed, and that the very best schemes would work only until someone figured out a clever way to foil them. RCA's Jim Gibson reminded us that possibly the only technical solution would be to make the cost of the recorded medium (compact disc, LP, or whatever) cheaper than the copying medium—certainly an unlikely happening.

Committee chairman Bart Locanthi reported on the general failure of various blank-tape tax plans. It was said that when such a tax was about to be added to blank tapes in Germany, the people changed the government and the tax never took effect. (Hm, maybe it *is* worth a try.) In Sweden, blank tape sales soared just before a similar tax was imposed, then dropped to almost-zero as the tax went into effect.

For whatever it's worth, we'll throw in our (tax-free) two cents. Stealing is a crime. (How's that for getting to the bottom of things fast?) If you'd like to have something (a record, a computer program, or whatever), you can avoid the criminal aspects by simply paying for it. Of course, if you can rip-off a copy from a friend...

Will a tax help stamp out this sort of crime? What do you think? We think a software tax is just another form of legalized crime, unless you're naive enough to really believe the tax you pay will eventually find its way back to the author or artist whose creation you are swiping. And besides, you don't have to be terribly bright to figure out that a tax to protect every author/artist from every lost sale would simply have to be enormous. And, you would no doubt have to report just what it was you were planning to copy, so the proper person could receive your tax. And, of course, you would have to promise never to make a new copy of something else on the same software. Otherwise, there would have to be a re-copying tax. In other words, you can't eliminate one crime by legislating another one.

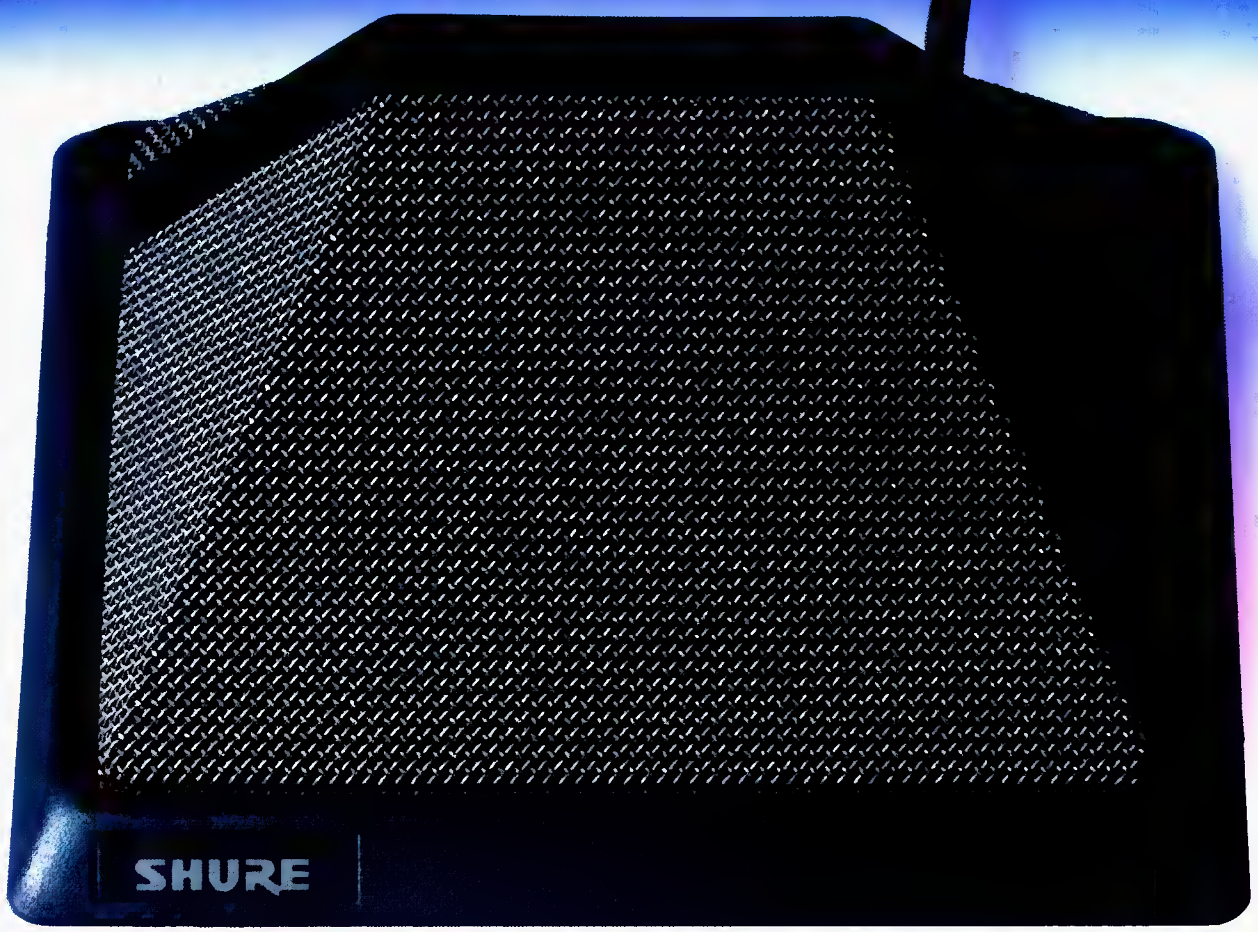
People have been abusing various sources for a very long time now. Guns, booze, nuclear power, drugs, and now magnetic media—all have been regarded with some amount of disfavor, as though the medium was the monster. (Do you suppose that after all these years, Pogo is still the only one who has truly seen the enemy?)

We seem to have painted ourselves into a corner, and there's no easy way out. Copying appears on its way to becoming a national pastime. There is no technically-infallible way to thwart copying (nor—as Dr. Stockham puts it—is there a way to convince *record executives* that there is no technically-infallible way to thwart copying). And finally, taxation schemes may well be a punishment worse than the crime.

We were going to come up with a brilliant solution to this mess, but it's getting late. So, we'll leave it up to you to work it out. We're going back upstairs to tape Nicholas Nickleby. JMW

INTRODUCING

THE SMART MIC* SYSTEM



The world's first and only Automatic Microphone System. From Shure.

*Microphones and Intelligent Circuitry

A trouble free microphone system to sol

The Shure Automatic Microphone System (AMS)–Complementary microphones, mixers and logic circuitry provide the key.

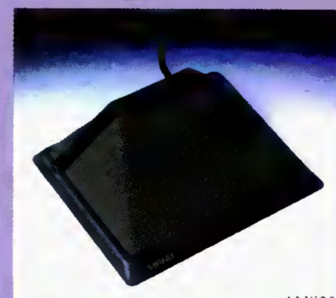
Now, the most aggravating problems brought on by multiple microphone installations can be solved. For the first time ever, Shure has combined unique microphone, mixer and logic technology into a dedicated, totally integrated system. A system where the microphones and mixer actually operate as one to provide the clearest, smoothest, most reliable automatic sound performance in the industry.

Smart Microphones– the newest angle on sound sensitivity.

Each microphone and mixer channel in the Shure Automatic Microphone System contains logic circuitry. This enables every microphone to act independently in the system when turning on or turning off. And, each microphone actuates only when addressed within a specifically tailored 120° acceptance window. Sound originating outside a microphone's acceptance window will not, in any way, actuate that microphone. In addition, each microphone continuously analyzes its own local acoustic environment allowing each channel to adjust itself independently as audio conditions change. This unique feature assures quick and easy set up by eliminating time-consuming sensitivity or threshold adjustments.

Two microphone styles– an edge in design and versatility.

The surface mounted AMS22 Low-Profile Microphone features a revolutionary design–a look so unintimidating, even a first time speaker won't shy away from it. The sleek AMS26 Probe Microphone is, in every way, the technological equal of the Low-Profile, and can be mounted on a table, floor stand, or gooseneck.



AMS22

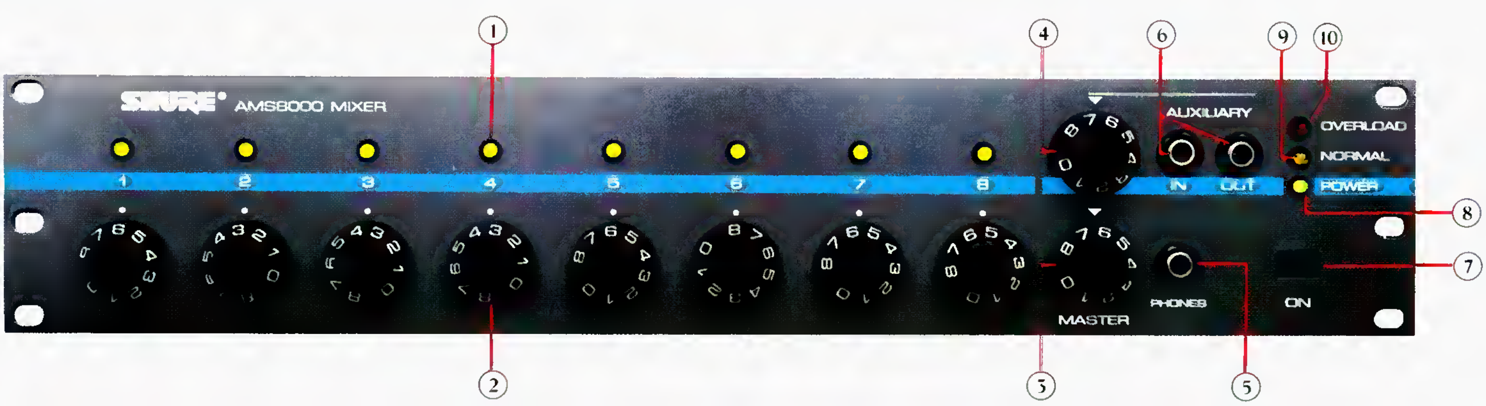
AMS26

A system that works together to produce outstanding sound performance.

Advanced microphone, mixer and logic technology enable the AMS to turn on to the sound source quickly, quietly and automatically–and turn off with a smooth whisper. From beginning to end–no clicks, pops, noise "pumping," or missed syllables. And, when more than

AMS8000 Mixer

1. **Channel LED Indicator:** Lights when a microphone turns on.
2. **Microphone Gain Control:** Controls channel volume only. The click-off "O" position is a "channel disable" feature. This eliminates unused microphones from the automatic gain compensation circuit.
3. **Master Control:** Controls output level of the combined microphone and aux input signals.
4. **Auxiliary Input Control:** Determines the level of the aux input.
5. **Headphones Output:** 1/4" phone jack, suitable for most stereo or mono headphones.
6. **Auxiliary Jacks:** Allows convenient connection to a tape recorder for playback or recording.
7. **On-Off Switch:** Push button controls power.
8. **Power-On LED:** Lights when the power is on.
9. **Normal LED:** Lights when the output level rises above approximately –20 dBV.
10. **Overload LED:** Flashes when the Line/Mic, Aux, and Phones outputs approach clipping.



e the most troublesome audio problems.

one microphone is actuated simultaneously, the system provides automatic gain compensation to prevent the annoying problem of feedback.

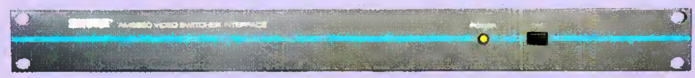
What's more, AMS operation is so easy and automatic, an operator's only concern is adjusting the individual volume controls. Setup is also quick and easy. And, there's never a need for repeated readjustment after the system is installed.

AMS—the system with a built-in future.

Every Automatic Microphone System incorporates advanced logic terminals which provide unprecedented flexibility for expanding the system's capabilities. These special capabilities include:

- *Privacy or cough button.*
- *Chairman muting.*
- *Channel priority—lets one speaker override another.*
- *Filibuster capability—allows only one microphone on at a time, to prevent interruptions.*
- *Zone loudspeaker muting.*
- *Remote channel indicators.*

The AMS system can even be connected to a computer programmed for even more sophisticated control operations. And for large gatherings such as symposiums or congressional meetings, AMS mixers (both 4 and 8 channel available) can be easily linked to effectively control over 200 individual microphones. In addition, when connected with the optional microcomputer-based Shure AMS880 Video Switcher Interface



component, the AMS will control commercially available video switchers. In doing so, television cameras will automatically follow the microphone channel activity to visually monitor prescribed areas.

Reliability—on location.

The Shure AMS is so technologically advanced, its conception marks the beginning of a sound revolution wherever speech related, multi-microphone systems are employed:

Churches. Shure's AMS microphones turn on and off smoothly and quietly—greatly reducing muddy sound and feedback.

Courtrooms. A direct output on each channel easily connects to a multi-track tape recorder to provide a word-for-word account of the proceedings.

Meetings and Conferences. The optional filibuster mode prevents interruptions by allowing no more than one microphone to be on at any time.

Teleconferencing. AMS can reduce the distant, "barrel" effect caused by room reverberation.

Security. When an intruder's sounds activate an AMS microphone, a television camera can automatically switch to visually monitor the trespassed area.

Broadcasting. Accurate mixing is guaranteed during fast-paced news and talk shows.

(11) **Off-Attenuation Switch:** Determines the amplitude of microphones that are not turned on. Keeping all microphones slightly on at all times contributes to smooth audio operation. Attenuation may be fixed at a recommended level of −15, or adjustable over a wide range.

(12) **Link Jacks:** Links up to 25 AMS mixers together to provide an input capability of as many as 200 microphones.

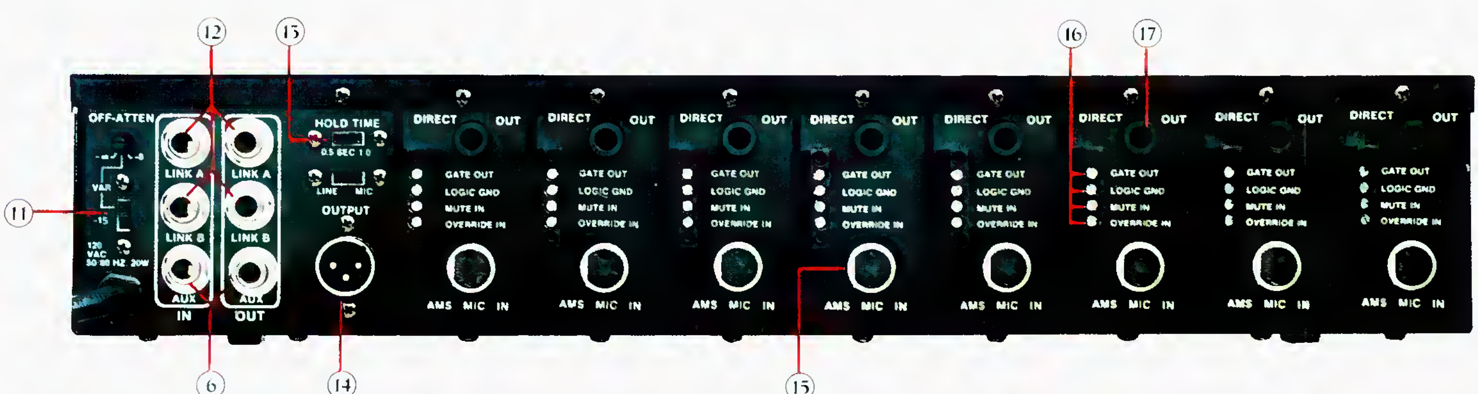
(13) **Hold Time Switch:** Determines how long a microphone stays on after the speaker stops talking. This delayed turnoff keeps the microphone on during brief pauses in speech.

(14) **Line/Mic Level Output:** Provides the combined gated microphone and non-gated aux input signals.

(15) **AMS Microphone Input:** The AMS mixer is supplied with either four (AMS4000 unit) or eight (AMS8000 unit) microphone inputs.

(16) **Logic Terminals:** Connections to these terminals allow the AMS to perform a variety of sophisticated functions.

(17) **Direct Output:** Provides a non-gated signal from that channel's microphone.



Specifications

AMS4000 & AMS8000 Mixer

Output Level:

INPUT	OUTPUT					Input Clipping Level at 1 kHz
	Line	Mic	Aux	Direct	Phones	
Microphone Input (72 dB SPL in)	+15.8 dBV (+18 dBm)	−34 dBV	+17 dBV	−56 dBV	−4 dBV	128 dB SPL
Aux Input (−22 dBV in)	+15.8 dBV	−34 dBV	+17 dBV	—	−4 dBV	+7 to +20 dBV*

*Depending on Aux control setting.

Frequency Response:
Aux Input to Outputs: 30 to 20,000 Hz ± 2 dB
Mic Input to Outputs: 70 to 20,000 Hz ± 2 dB
(controlled low-frequency rolloff below 50 Hz)

Outputs:

OUTPUT	IMPEDANCE		OUTPUT Clipping Level
	Designed for Use With	Actual (Internal)	
Mic	150Ω balanced lines	1Ω	−34 dBV
Line	600Ω balanced lines	150Ω	+15.8 dBV (+18 dBm)
Aux	10k or greater	2.2k	+17 dBV
Direct	10-50k unbalanced mic circuit	900Ω	0 dBV
Phones	200Ω	2.2k to tip 2.2k to ring	−4 dBV

Equivalent Input Noise:
27 dB SPL A-weighted, with AMS26 Probe Microphone

Distortion:
THD 0.35% or less, 30 to 20,000 Hz at +15 dBm output
IMD 0.5% or less up to +15 dBm output

Gating:
Attack Time: 4 msec
Hold Time: 0.5 or 1.0 sec (switchable)
Delay Time: 0.3 sec after Hold interval

Off-Attenuation:
Fixed: −15 dB Variable: −∞ to −8.5 dB
(Single mixer; attenuation increases as additional mixers are linked)

Operating Voltage:
105-132 Vac, 50/60 Hz, 20W. Can be rewired for 210-264 Vac, 50/60 Hz, 20W.

Dimensions:
Height: 89 mm (3½ in)
Width: 483 mm (19 in)
Depth: 298 mm (11¾ in)

Weight:
AMS8000: 6.6 kg (14 lb 8 oz)
AMS4000: 5.8 kg (12 lb 13 oz)
AMS8000 (pkgd): 8.4 kg (18 lb 8 oz)
AMS4000 (pkgd): 7.6 kg (16 lb 13 oz)

Certifications: Listed by Underwriters Laboratories, Inc.

AMS22 Low-Profile Microphone

Type: Condenser (electret bias)
Polar Pattern: Hemi-Cardioid
Acceptance Angle: Microphone turns on for sounds within 60° ± 10° of front axis
Output Level: (Open circuit voltage at 1KHz)
−47 dB typical (0 dB = 1V/μbar)
at AMS mixer direct output
Noise: 20 dB equivalent SPL typical, A-weighted
25.5 dB equivalent SPL typical, weighted per DIN 45405
Cable: Non-detachable 6.1m (20 ft), 2-conductor, shielded, with 3-pin professional audio connector to mate with Cannon XL series, Switchcraft A3 (Q.G.) series or equivalent connectors
Case: Black plastic base and brown steel-mesh screen with black trim

Dimensions:
Height: 31.8 mm (1¼ in)
Width: 88.9 mm (3½ in)
Depth: 76.2 mm (3 in)

Weight:
Net: 174 grams (6.1 oz) including cable
Packaged: 360 grams (12.6 oz)

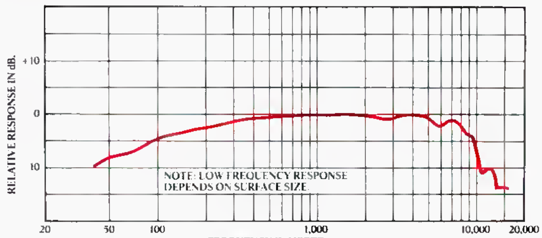
AMS26 Probe Microphone

Type: Condenser (electret bias)
Polar Pattern: Cardioid
Acceptance Angle: Microphone turns on for sounds within 60° ± 10° of front axis
Output Level: (open circuit voltage at 1KHz)
−54 dB typical (0 dB = 1V/μbar)
at AMS mixer direct output
Noise: 27 dB equivalent SPL typical, A-weighted
32.5 dB equivalent SPL typical, weighted per DIN 45405
Connector: Three-pin professional audio type to mate with Cannon XL series, Switchcraft A3 (Q.G.) series, or equivalent connector
Case: Brown vinyl-enameled brass handle with brown steel-mesh grille

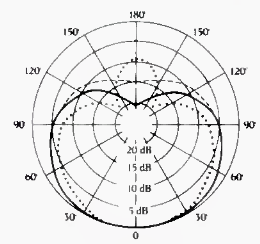
Dimensions:
Length: 144 mm (5 21/32 in)
Diameter: 35.9 mm (1 13/32 in)

Weight:
Net: 127 grams (4.4 oz)
Packaged: 366 grams (12.8 oz)

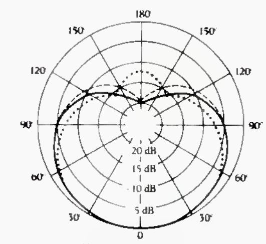
Furnished Accessories:
Windscreen RK229WS Swivel Adapter A57D



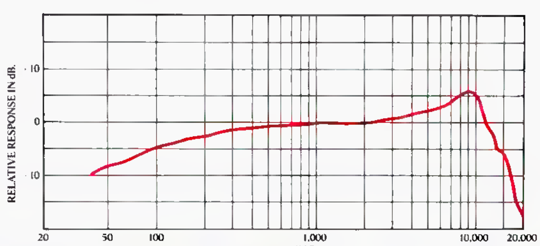
AMS22



AMS22



AMS26



AMS26

AMS880 Video Switcher Interface

Write or call for specifications.

SHURE®

SHURE BROTHERS INC., 222 HARTREY AVE., EVANSTON, IL 60204

Printed in U.S.A. 12/82 AL721

JAMES F. RUPERT and MICHAEL ROBERTS

Keep On Trackin': Barry Ainsworth and Mobile One

So you can't make it to the recording studio? No problem— the studio will come to you.

IMAGINE, IF YOU WILL, booking a recording date and then finishing the arrangements by saying, "Is it okay if I drive the studio over to your place on Thursday around 'sevenish'?"

While this statement may sound alien to some, it spells business as usual for Barry Ainsworth and the crew of Mobile One, Europe's largest mobile recording facility. Beginning his audio career as an assistant engineer with Pye Studios in the early 60's, Barry later designed and built the now legendary Sarm Studios and became a founding director of the facility. One of the world's most successful studios in the early 70's, Sarm became the first 24-track and computer mixing studio in Europe.

Having worked with such diverse artists as Long John Baldry, The Foundations, Spencer Davis, Deep Purple, Jimi Hendrix and Donovan, Barry Ainsworth also has the distinction of being responsible for the engineering of no less than nine top ten recordings within one week's time (definitely our candidate for this month's "Golden Touch" award)!

In 1975, Barry and Pye associate Bill Foster began the design and construction phase of Tape One Studios, with a disc cutting room following in 1976. With Tape One now able to boast of three disc cutting rooms, two mastering rooms and digital mastering and editing facilities, Barry confines his activities basically to Mobile One operations.

db tracked down Barry on location at the University of London, where Mobile One was being set up to record the audio portion of a videotaped jazz concert featuring Slim Gaillard, Kenny Clarke, Kai Winding and Johnnie Griffin. Past Mobile One artists have included Yes, AC/DC, The Kinks, Rick Wakeman, Dr. Hook, Rush, Genesis and Supertramp, whose LP *Paris* earned not only a gold record for the group, but an Ampex Gold Reel Award for Mobile One!

Operational for the last four and one half years, the 46-track facility began as a joint production between Barry and Radio Clyde, a commercial radio station in Scotland. While other mobile recording facilities existed at the time, Barry envisaged a true state-of-the-art "rolling studio" rather than just a conveniently transportable recording setup. Over a year and a half from conception to christening, Barry told us: "It took almost 12 months to build, but it's been on the road since then."

INSIDE MOBILE ONE

Mobile One is the first Hidley-designed mobile outside of America. The heart of the recording system is a specially rebuilt 400 series MCI 36 in/36 out console, complimented by a Triad 16 in/16 out auxiliary console which allows up to 52 input channels and 52 outputs. "We deal a lot with engineers that come in specifically for one job," Barry remarked. "One of the criteria when I built this truck was that it could be operated very quickly by anybody. The desk is quite a simple one. The reason for this is that I wanted a desk that an engineer could sit down in front of and not be faced by a million and one knobs!"

Although Mobile One normally offers two 24-track MCI recorders linked through SMPTE/EBU time code synchronization, Barry is quick to point out that the truck can be set up for digital recording on special request. "We have the Sony PCM 1610 2-track and digital editor on hand, but we will arrange for whatever system the customer wants.

"We can hire a 24-track machine direct from Sony or a 3M machine from the Roundhouse Studios here in London."

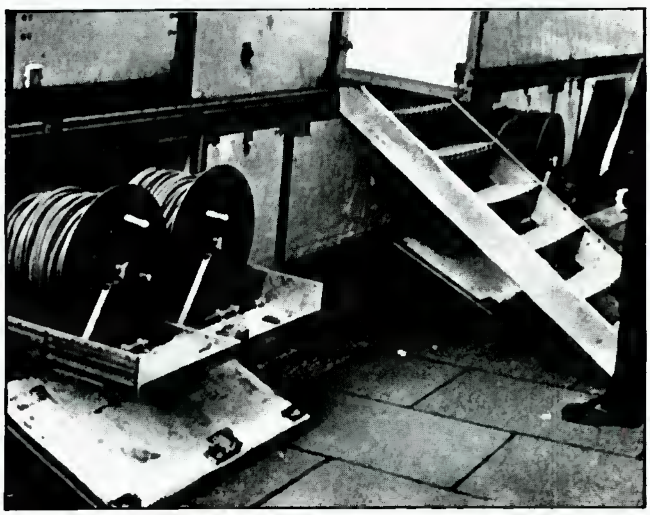
Any customer preferences? "We do quite a lot of classical work for CBS records using the 3M system," Barry said, "and quite a bit of rock and roll with the Sony. From there we can mix it down to two track digital with our own equipment."

When Barry was asked for his personal reaction to digital recording, his face spread into a wide grin as he answered. "I like it," he beamed. "I mean, it's obviously the way the business is going and you've just got to go along with the business. That plus the fact that we've not had many problems with digital technology, thank goodness.

"We're doing quite a lot of digital work at Tape One Studios. There're one or two companies producing CD format tapes for transfer to CD discs, and we're producing the initial tapes for cutting. At this point we're not even planning a digital cutting room because of the expense."

TAPE ONE STUDIOS

What Tape One does offer is a computer-controlled Neumann VMS 80 disc cutting lathe and Neumann SP79 cutting console. The lathe is fitted with an SX74 cutterhead that



Microphone snake cables mounted on revolving spools

James Rupert is an engineer and an acoustical design consultant. Michael Roberts is a radio management consultant.

A QUESTION OF STANDARDS.

48kHz. VS. 44.1kHz.

The recent explosion of digital audio systems has also raised questions about digital technology. What follows are some answers from Mitsubishi Electric to questions we find most often asked about one aspect–sampling rates.

Question: What is meant by the term, "sampling rate"?
Answer: In digital audio recording and playback systems, the sampling rate is the speed at which the conversion from analog to digital takes place. In theory the faster this "sampling" occurs, the wider the frequency response of the system will be. This is because more samples per second are used to digitally describe the sound.

Question: Why have Mitsubishi Electric and most other manufacturers of professional digital audio systems used the rate of 48kHz. as their "common sampling rate"?
Answer: It was important to establish a "common sampling rate" to allow the simple transfer of digital audio recordings to other media, such as digital playback systems and broadcasting lines. Again, in theory the higher the sampling rate the wider the frequency response of the system will be. Since the rate of 48kHz. is roughly 9% faster than the 44.1kHz. rate used on some other recording systems, the Mitsubishi recorders can reproduce frequencies above the normal upper limit of 20kHz. Secondly, since this upper cutoff frequency is higher than 20kHz., it allows for the design of simpler low-pass filters, thus minimizing any phase-shift characteristics throughout the audible spectrum. Finally, since digital audio broadcasting lines already established in Japan and Europe utilize a sampling rate of 32kHz., the transcoding of professional digital mastertapes is relatively simple.

Question: Then why use the lower 44.1kHz. rate at all?
Answer: This sampling rate is the one used on all Compact Disc playback systems and represents an excellent choice for the playback of those frequencies in the 20Hz. to 20kHz. range. The lower rate does not allow for some of the features that professional recording systems require, however.

Question: Then can digital mastertapes produced on the 48kHz. professional systems be made into Compact Disc software?
Answer: Certainly. The procedure is relatively simple and is performed at the CD pressing plant.

Question: Are there any other advantages to using the 48kHz. rate on professional digital audio systems?
Answer: Yes. One feature important to the creative process of making and recording music is that of variable-speed operation (VSO). To alter the pitch of digital mastertapes it is necessary to likewise alter the sampling rate slightly. To perform this VSO and still maintain the high-quality specifications of digital audio, the system must use the rate of 48kHz.

Quality. Versatility. Specifications.
At Mitsubishi we have the same commitment you do.

MITSUBISHI
DIGITAL AUDIO SYSTEMS

THE BEST ANSWER FOR PROFESSIONALS.

Circle No. 27 on Reader Service Card

operates in conjunction with a pair of SAL74 drive amplifiers providing 500 W peak power per channel. As with all Tape One and Mobile One layouts, facilities design was by Barry Ainsworth and Bill Foster.

Initial demand for Mobile One was centered basically in Continental Europe, with the truck zigzagging its way through France, Germany, Italy, Switzerland and Spain. Currently the amount of work has grown to the point that Barry and engineers Andy Rose and Tim Wybrow find it difficult to ever leave Great Britain.

The massive insulation within the walls of the truck has served for more than just sound isolation. Once, while recording during an ice storm in Yugoslavia, the Mobile One crew was forced to use heat lamps to free cables frozen to the ground during the session. Yet within the truck the air conditioning was running on high to keep temperatures down to shirt-sleeve comfort.

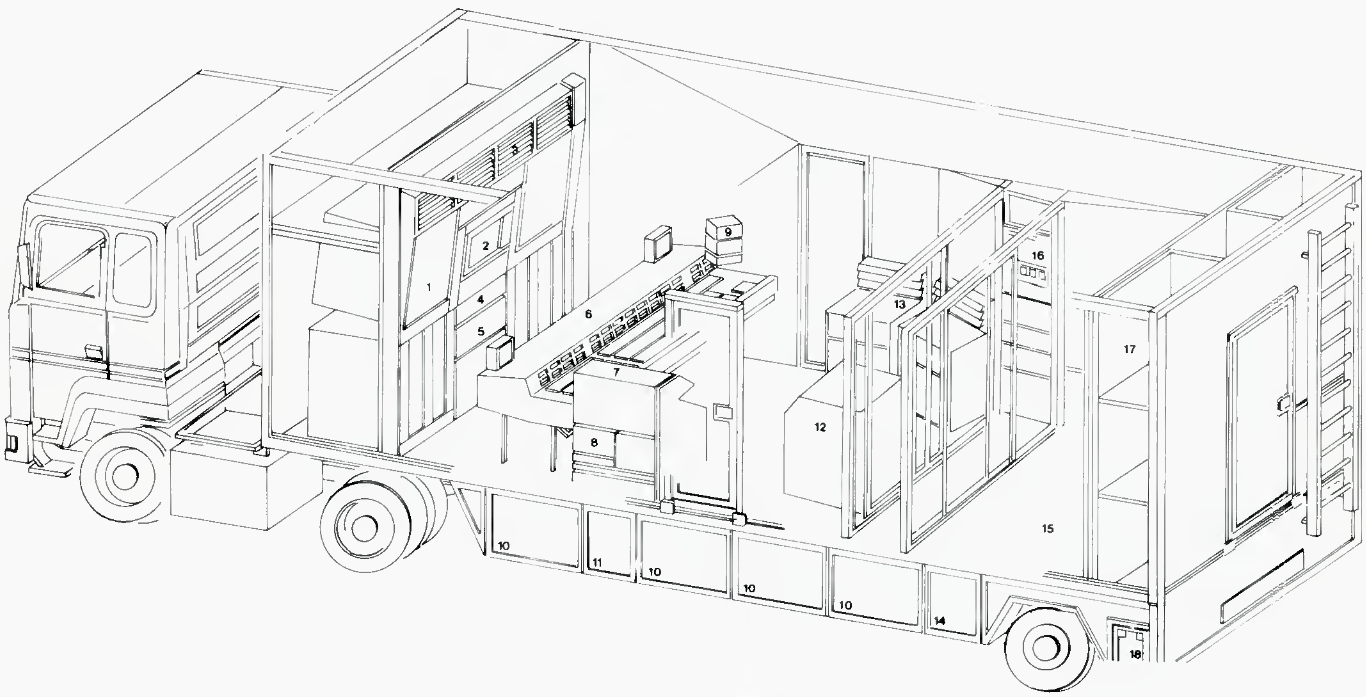
Added to features like a self-contained generator for independent power and a mini studio in the rear of the truck, the 14-meter mobile has its own radio telephone which doubles as a security system. "If anyone did break in they'd be in for a bit of a surprise," Barry explained. "In addition to triggering some enormous bells, the radio phone is connected to a little tape that tells the operator to call the police that we're being burgled. Actually, I think the bells would scare someone off more than anything, but the entire system is connected to every locker and door on the truck."

WAIT, THERE'S MORE

The success of Mobile One has led quite naturally to the introduction of Mobile Two, a two-track facility featuring a Neve console and Studer recorders. Manned by engineers Ray Prickett and Alan Stagg, Mobile Two is designed for portability and the ability to be transported to even the smallest locations.

With 70 to 80 percent of Mobile One's bookings being video-connected, a Mobile Three video studio is presently in the conceptual stage. Is video to be the recording wave of the future?

"Yes," Barry replied. "We often find ourselves doing audio for a record and a video production simultaneously. Now that



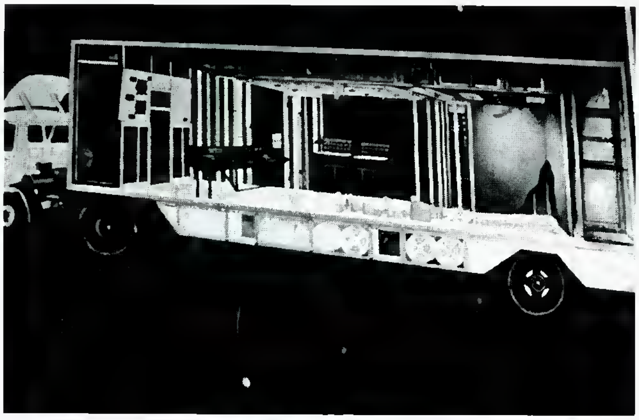
1. Eastlake monitoring
2. Color video monitor
3. Air conditioning
4. & 5. Amcron amplifiers and White equalizers
6. Mixing console
7. Auxiliary mixing console
8. SMPTE/EBU Time code synchronizer and tape recorder remote controls
9. Effects Rack—includes EMT Digital Echo; Master room reverberation unit; Comprehensive "Scamp" effects rack (compressors, limiters, etc.); UREI limiters; Eventide Harmonizer, and UREI Parametric Equalizer
10. Cable lockers
11. Audio input
12. 2/4 track and cassette recorders
13. Two 24-track tape recorders
14. Mains input
15. Small studio
16. Power control and switching
17. Pressurized storeroom
18. Microphone stands

Mobile One Line-Up

- MCI recording console 36 in/36 out, can be used with Triad on-board console (16 in/16 out for a total of 52 in/52 out.
- 2-MCI 24-track 2" recorders. Can be tied together with SMPTE code generator to achieve 46 available recording tracks.
- MCI 2tr/4tr mixdown machine. Sony 2tr digital recorders available on request.
- 48 channels Dolby A noise reduction.
- EMT digital and Master Room echo units.
- Eventide Harmonizer, UREI Limiters, UREI Parametric Equalizers.
- AKG, Beyer, Neumann, Sennheiser and Shure Microphones.
- Hidley-designed JBL/TAD monitoring system, with Auratone Mini Monitoring secondary system.
- Full video (remote-controlled) monitoring system w/color monitor.
- Radio Telephone
- Independent power generator (when required).
- Full synchronization with external videotape recorders.
- Full air conditioning.

television programs are getting cheaper to make, we're doing a lot more recording with video cassettes and discs in mind. As a matter of fact, with so much of our work involved in television, audio has almost become a spin-off.

"At one time you could sit back and expect people to come to you with projects. You've got to go out and get it now. You can't sit still, you've got to keep working at it. Marketing is far more important today than it has been the past few years."



Cutaway view of Mobile One

Interior equipment layout

Part of that marketing effort recently resulted in a unique first for Mobile One: A 24-hour turnaround on an LP release. Barry and his crew recorded a live gig for "Toots and the Maytals" at Hammersmith Odeon and on the following day had the material mixed, mastered, pressed into discs and available in local stores. (The management suggests that such an effort should qualify for the Guiness Book Of World Records, and they will certainly receive no argument from these authors!)

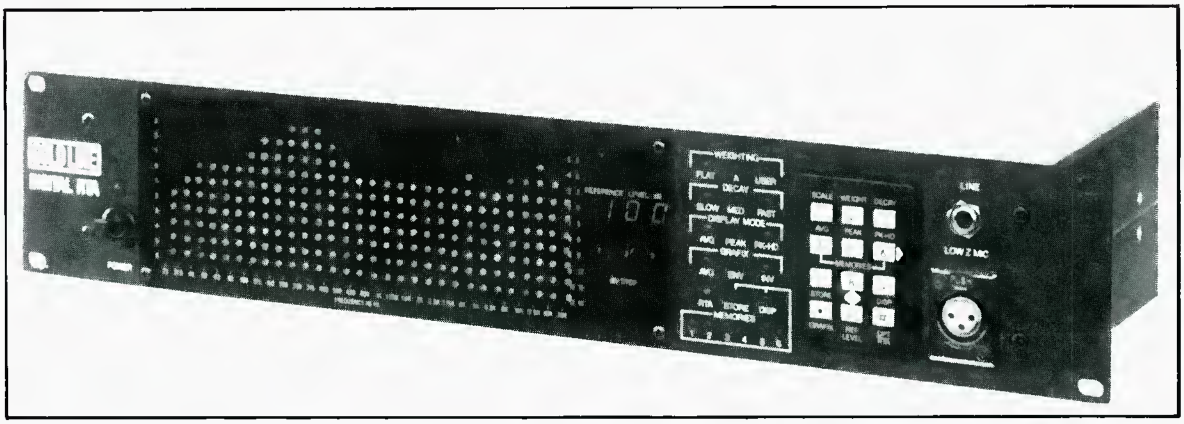
With as impressive a list of credentials as Barry and Mobile One possess, what lies ahead in the future? "I don't know," Barry chuckled at the question. "I could well still be running this, but whether I'll be actually in the truck is another story. I enjoy it totally at the moment. But who knows quite frankly what next week will bring? These days, if you get through a week successfully you're doing quite reasonably well!"

Quite reasonably well indeed, Mr. Ainsworth! ■

J. MARK GOODE

db Test Report

The Gold Line 30 Real-Time Analyzer



If you do a lot of rock gigs and you're looking for a real-time analyzer for your P.A. system, the Gold Line 30 may be just what you're looking for. The Gold Line 30 is a digitally controlled, one-third octave analyzer with a substantial list of additional features. It's designed for quick, easy operation, making it a real time saver on the road.

THE FRONT PANEL

The Gold Line 30 is a standard 19-inch rack mount size with a durable front panel. The controls for the device are operated by keypad, eliminating knobs and switches that might become loose or break off on the road. The LED readout is a thirty-by-ten dot matrix covered by a thick plastic shield for rugged use. An additional set of LEDs to the right of the dot matrix provides information on status and operating conditions of the RTA. The keypad consists of minimum-travel buttons to prevent damage during heavy use. The only design weakness on the unit's exterior is the power switch, which is a plastic toggle. A power button would have made the front panel virtually indestructible on the road. Overall, it is a clean and neat easy to read design.

THE MANUAL

The Gold Line manual is short and easy to read. It provides a fast and efficient reference for operating the unit but supplies a minimum of technical data. The technical references are only one page in the back of the manual and give information regarding impedances and levels of inputs and outputs. Although the filters are described on the second page, no information is contained as to specifics in filter Qs, out-of-band rejection, or analog/digital information. I would also like to see some flow-chart information since the filters are an analog/digital hybrid and the unit is controlled digitally. For example, are the readings on the front panel digital representations of a sampled signal, or are they only the outputs of the filters digitized? Are the

University of Miami student J. Mark Goode is currently chief engineer at the school's recording studio.

sound pressure levels read on an already-sampled signal, or is the measurement done on an analog signal and the output digitized? In its use as a semi-pro device on the road, this information is probably not needed, but in analytical work of a more demanding nature, this information could be vital in determining the amounts and natures of errors made in the digital readout. As noted, the operator's manual has the advantage of being short and simple so that the user can understand and operate the RTA very soon after receiving it, but a supplemental technical manual of about equal length would be helpful.

TECHNICAL SPECIFICATIONS

The technical specifications are quite adequate for the general road gig. The Gold Line 30 functions are handled by an Intel 8085 microprocessor. This allows clock rates of greater than 6 MHz, and sampling for the 300-dot matrix once every two milliseconds. The filters are first-order switched-capacitance filters. With their center frequencies controlled by clock pulses, the filters are immune to drifting caused by temperature changes or age. In sweeping the input with a sine wave I found at most a 1 dB variation in the LED reading. One dB is the smallest change readable on the LED matrix. Gold Line claims 0.5 dB, which is entirely believable. The filters were very well matched with no visible variation as I went up and down the frequency spectrum. One problem I encountered at higher frequencies was an incorrect reading on the lower frequency filters. The problem was at a maximum as the parameters of the input wave were changing (amplitude and frequency). With an input of 10 kHz, which varied by 20 Hz each way, a reading at 50 Hz and 100 Hz occurred at only 3 dB less than the input signal itself. This is similar to aliasing. Since noise is the thing to be measured—and noise, by definition, is constantly changing—this effect could present a problem.

The Gold Line 30 is equipped with a pseudo-random pink noise source with appropriate gating for RT60 readings in the future. When the unit analyzed its own pink noise it was flat, ±2 dB, with a 1 dB hump at 800-1000 Hz. Its output is an unbalanced ¼-inch phone plug and the pink noise is at a nominal level of 700 mV. The inputs are line level high impedance with a ¼-inch phone plug which disengages the balanced microphone input. The mic input (XLR) has phantom power available at the jack in the order of twelve volts. The input can accept any calibrated, low-impedance microphone of the kind used for pink noise analysis, but Gold Line recommends the AKG 451 E series or Gold Line's own MK-30 microphone, made for the unit and available as an option. The RTA can also be ordered with the inputs on the rear panel. Since the line level is unbalanced and high impedance, some care must be taken not to run long lines to the input from an unbalanced output. I occasionally had trouble with 60 Hz hum appearing on the display.

Though the unit is designed as a rack-mount device, it generates quite a bit of heat after long use, so some attention to ventilation is necessary. The back of the RTA has large heat-sink fins which do the job adequately, but the top of the unit runs warm and does need ventilation.

By using a digitally-controlled system, the digital representations of the filter outputs can be stored for later use. The RTA comes with six memory locations which can store the outputs of the LED display and sum them as the need arises. This allows the user to take readings at many places in the listening environment, sum them, and obtain an average response for the room, rather than for just one location. The most impressive aspect of the Gold Line 30 RTA is the list of options which are now available, or should be soon. One option, contained as standard equipment, adjusts the reference level of the display to the average of the individual bands. This allows storing values already adjusted for overall sound pressure levels. The sum of memory locations can then be independent of the total level at any one point. The addition section can hold as many as ten curves to be added. This gives the user the ability to weight them, giving preference to one location over another. By filling up the memory locations, summing them and storing the results, repeating the operation can provide as many as 21 different readings. The summing procedure requires that the scale of the display, the weighting of the curve, and the mode of the display be consistent throughout the set of curves to be added.

The scale of the display is adjustable to 1, 2, or 3 dB increments, giving a total range of 10, 20, or 30 dB respectively. The 3 dB setting is generally sufficient for rooms with no extremely bad responses, and the 1 dB scale works very well for the final fine tune. Over-ranges are indicated by the top two LEDs in peak mode and the disappearance of the bottom LED in average mode. Over-range in average mode can sometimes be tricky to read at a single glance. The thirty filter bands correspond to ISO center frequencies for ease of equalization.

The response of the RTA can be weighted—either flat or standard-weighting. A user-defined weighting is available as an option if the parameters of the curve are given to the Gold Line people when ordering the unit.

The display can be made to read either peak or average response. The peak response is displayed with points, while the average response is displayed as a bar graph. The peak-hold function works on either mode but cannot be used if the functions are to be summed later. In the event of inconsistency, the display will signal ERROR with the offending functions flashing. The error messages are clear and free from ambiguity. The display also has three set rates of decay: 3.5-, 7-, and 21-dB per second.

An optional battery pack can be ordered and is connected to the back of the unit with a nine-pin Molex plug. The battery pack is rechargeable and is operated by a switch on the back of the RTA which selects between the battery pack and line voltage. The Gold Line 30 can be ordered to accept line voltages of 110 or 220 VAC and 50 or 60 Hz. These cannot be changed on the unit itself by the user. The European-option units are designed to meet VDE specifications.

The options—some available now, others in the future—should help make the Gold Line 30 very adaptable to specific needs. Options which are now available, along with the user weighting, include the battery pack, the MK-30 microphone, an involatile memory (unlike the standard memory which erases itself in the event of a power failure), printer interface, and an inverse function.

The involatile memory is a 2k RAM with a lithium backup which can hold as many as 256 additional curves. Compared to the basic unit's six available (volatile) memory locations this option increases user power significantly.

The printer interface uses the same MX-80 printer used by the IBM Personal Computer. These options are not designed to be user added and so must be ordered and installed by Gold Line.

The inverse function takes the reflection of the sums about the reference level so the equalization settings can be matched to the visual display. The inverse function is a software change that requires no additional hardware but must be specifically ordered. This is puzzling since it would seem that the manufacture of the system ROM would be the same expense with or without the added function.

Three expansion ports and an extra PROM location contained in the unit allows more complicated functions to be added in the future. Some of the proposed options include RT60 analysis, computer plotters, and CRT interfaces. These are not available now but will be eventually, along with several music options.

The most important thing about a real-time analyzer designed for road use is its performance on the road. After spending much time with signal generators and test equipment, I gave the Gold Line 30 out for a road test: it performed quite well. Installation of the unit took about 15 minutes (working alone). The unit takes about 20 seconds to power up as it flashes messages on the display: AUTO CAL READY. In this procedure the Gold Line reads the outputs of all the filters and compensates for any DC biasing. It then falls into RTA mode, flat, average, slow decay, and, at 100 dB SPL, reference level. AUTO CAL can be done at any time by pressing OPTION and 9 on the keypad.

The unit always goes back to RTA mode, which is the starting point for calling any special feature.

Since the microphone is not in a fixed place, some assistance is necessary to take readings. A great deal of time can be saved if one person moves the microphone while someone else records the readings. Panel readings can be taken and stored in a matter of seconds and then summed when a sufficient number have been taken. Using the two-man system, preliminary readings can be taken all over the room in about five minutes—less in a smaller room. With the display set on the 3 dB per division range, rough equalization settings can be made quickly. By changing the display to progressively smaller divisions, fine tuning can be achieved in half an hour. Although I did not have a user curve option on my analyzer, use of this option would quickly define the room to any response the user decides. The user curve must be set at the factory, however, and cannot be changed at the concert sight. The entire process is quick and efficient. It renders a good room response average, unaffected by problems with standing waves and the like inherent in a single-point sample.

One problem I did encounter was that the lowest reference setting for the real-time analayzer was 50 dB SPL. This setting does not allow for the analysis of the ambient environment, and the line level corresponding to this reference is too low for quiet passages. This limits the device to P.A. room tuning applications and makes it unusable for extreme analysis situations. The aliasing error discussed earlier led to some error in the low-frequency readings which also limits the device in its use under adverse conditions. The resolution is somewhat restrictive in the analysis of specific harmonics of individual instruments, and the maximum range (30 dB) is a good bit less than that of playing instruments. The filter design (first-order filters) eliminates the use of any range higher than 3 dB per division since this scale already shows a lot of filter skirting. The automatic level control which sets the average level of the LED display to the 0 dB center line updates only twice a second, which makes it too slow to respond to live music. Live music usually forces the display out of range to the point of being useless. the versatility of the Gold Line 30 among P.A. tuning and road trips is excellent, but its use as an analysis tool is therefore somewhat limited.

CONCLUSION

To summarize the advantages of the Gold Line 30 Real Time Analyzer: it has adjustable ranges of 10 dB to 30 dB and has adjustable weightings of flat, A, or user-specified (optional). It also has six memory locations, adds them, gives wide-band sound pressure level readings, and has a long list of options to afford some degree of custom designing. It also comes with a one year limited warranty. Options now available are the battery pack, non-volatile memory, printer interfaces and user curves. Future options, to help insure that it is technically up-to-date, include RT60 analysis, CRT interfaces, and computer plotters.

The disadvantages include the sensitivity limits, filter design limitations and the lack of technical data in the users manual. Also, many options which will make the device truly outstanding are unavailable at the moment, such as RT60 analysis and CRT interfacing. And the device is slightly limited to P.A. type applications.

If you are doing a road gig and need a real-time analyzer, this is a good one. It's quick and easy to set up and use; in one session the user can easily become familiar with its operation. The human interface is well thought-out and the unit is tough and durable; the design is clean and attractive.

As we move from the analog to the digital world, digitally-controlled devices such as the Gold Line 30 will help to bridge the gap. Given the many available features (plus the ones to come)—and the modest price of the Gold Line 30 ($1695)—this unit bridges that gap quite nicely. Among P.A. engineers it will surely be a success. ■

SHERMAN KEENE

The Tale of the Telearte Recording Studio

Caracas is a wild and crazy place if ever there was one.

WHAT IT'S LIKE

IMAGINE THE CRAZINESS should some extra-terrestrial prankster remove all the stop signs, street signs, house numbers and police from the entirety of a large, well-behaved US city one quiet night. After a time, once the freedom from any kind of traffic rules (and traffic tickets) sank in, you might end up with a scene such as I found in Caracas. The "creative driving" styles, mandatory dents in every car, an unwritten traffic law stating: "Thou shalt change lanes every fifteen seconds whether you need to or not," hoards of death-wish motorcyclists and profuse numbers of "horn-beeping" areas completes the craziness. There is just one rule: When cars bump, both (or better yet, all) vehicles must stop right where they are until one of the (far too few) traffic police arrive on the scene. There is one law the drivers *do* obey—stopping right in the middle of main streets all over the city, regardless of the traffic tie-up consequences.

And then there are the bands of motorcyclists who, for kicks, have been known to drive up onto the freeways of Caracas going the *wrong way* and drive at breakneck speeds along the dashed lines between the oncoming cars at night with their lights *off!*

However, the Caracas weather is absolutely perfect: 85 degrees day and night, beautiful clouds drifting over head and nice humidity. Frogs beep a beautiful melody at night and sing in sync with each other, making a chorus of major and minor third chirpings. The senoritas are very lovely and the food (especially sea food) is extraordinary.

THE EXPERTS ARE CALLED IN

Into this spirited, energetic and crazy world enter the experts (*Los Expertos?*). Ultra precise, punctual and well-considered denizens of a totally different world—these experts know all about state of the art audio equipment. They were invited to Caracas to build a modern recording studio and install the best recording equipment available. They built the studio, installed the equipment, and then, work completed, they left.

Looking back into the studio. Notice the dome diffusers on the ceiling and the time-align studio playback system.

Sherman Keene is the author of Practical Techniques for the Recording Engineer.

HOW THE STUDIO CAME TO BE BUILT

Many moons ago, while at Best Audio in North Hollywood, Dave Brand was contacted by Dr. Armando Guia of Telearte in Caracas, Venezuela. He wanted to know if Dave would be interested in helping to build a very special studio in Caracas similar in many ways to Wally Heider's Studio Four in Hollywood. Dave agreed to help, and wrote a studio equipment and construction proposal thru Best Audio. The proposal named Chips Davis as acoustic and architectural consultant. Best Audio, with Sphere Electronics (under Dave Brand's supervision), designed the hardware system for the studio and specified how the studio was to be built.

The next chapter in the story consists of the entire order of equipment (a whole studio in boxes) sitting in storage in Caracas, waiting and waiting for the studio building to be finished. Things were getting bogged down due, in part, to exceptionally efficient Caracas customs officials. Did you know that it is not legal to ship a roll of multi-conductor wire to Caracas, but that a length of the same wire, once connectors are mounted, is legal? During this time Chips became more involved with the project and Dave less involved. The studio progressed—but very slowly.

Time passes. Dave Brand left Best Audio and formed HNE of North Hollywood (creators of the Westwood One remote recording truck) with Dave Farragher and Jim Seiter. HNE was called upon to come to Caracas and supervise the final "push" towards completing the recording studio. HNE sent Jim Seiter (an accomplished architect in his own right) to assist Chips in providing the necessary impetus to finish the project. Between the two of them, they actually got the studio finished.

A WORD ABOUT THE PRESIDENT

The president of Telearte, Peter Bottome, has a number of secret weapons for leaving his competition far behind. The latest is, of course, his newly completed recording studio. Another is his formidable fleet of personally-owned aircraft which includes a flawless Mustang P-51 (putts along at 440 knots!), a beautiful PB-Y (Catalina) flying boat (more on this later), a helicopter and some odds and ends of more modern aircraft including a twin turbo-prop Cessna Corsair (whew! what a plane). Last but not least is a more personable secret weapon: Peter's affable vice-president, Doctor Armando Guia.

Doctor Guia's task in the creation of the recording studio was to sift through the reams of written information describing the world's professional audio equipment and wisely select the best components, systems and people who could make it all come together in far-distant Caracas. I think Dr. Guia did a great job.

Together, the consultants and Dr. Guia selected: a Sphere Eclipse "C" 32/34 console; an Ampex ATR 124 multi-track recorder; ATR-102 and 104 recorders; 24 channels of Dolby

(M24), plus six 361 noise reduction systems; Urei Time-Align speakers and matching (servo) amplifiers for the control room and for the studio; one Lexicon 224 Digital Reverberator; one Lexicon Prime Time Digital Delay; one UREI 1178 and 2LA4 limiters; one dbx RM 160 dual limiter; an Eventide Omnipressor; one Eventide H949 Harmonizer; one AKG BX-20 reverb unit; one Echoplate reverb unit; two Valley People "Dynamite" processors; two RTS RIAA phono preamps; two Technics SL1800 turntables with Shure SC39ED broadcast studio cartridges; Yamaha instruments (electric bass and guitar); Yamaha instrument amplifiers; an Oberheim OBX, DSX and DMX; a Rhodes Chroma and a Yamaha acoustic piano.

DESIGN NOW, CONSTRUCT LATER

Beyond assisting in the selection of the above-mentioned odds and ends of high quality hardware, what the California and Nevada consultants actually provided for Telearte was a carefully tailored, integrated, de-bugged and operational studio in a "box" (a shipping box). All the rack equipment, for example—limiters, effects, power amps, etc.—were pre-wired by Best Audio with formed harnesses. What this meant is that Best constructed the whole Caracas studio on the floor of their North Hollywood factory, including all the multi-conductor cables and their multi-pin connectors, checked out the whole studio, de-pinned all the cables from their connectors and sent the finished (still pinned) cables and their connectors to Caracas in a box.

Masterminded by Dave Brand, this pre-wiring was carefully documented; "maps" of which rack positions each piece of gear was to occupy, wire number and label lists, etc. were prepared. When the box arrived in Caracas and all the goodies inside were finally hooked up according to the maps, diagrams and instructions, it all worked.

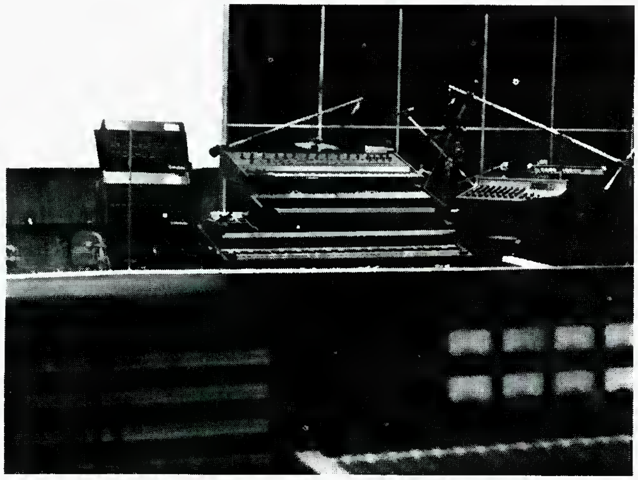
After the basic studio construction was completed, Chips Davis went to Caracas to oversee the finishing touches of the acoustic construction. Chips found that his blueprints had been realized into a very smooth sounding LEDE control room and a beautiful LEDE studio. Later, Chips was joined by Jim Seiter of HNE who designed and constructed the second LEDE (Live End/Dead End) studio in the world—Wally Heider's Studio Four. Chips built the first LEDE studio—in Las Vegas.

The "room sound" in the studio is the best I've ever heard, both in the room (in person) and on tape (through mikes). The control room acoustics are excellent and make the Urei Time-Aligned monitors sound fantastic. Chips' studio design looks good and works well, being both artistic and acoustically conducive to music making. Musicians can hear themselves and each other clearly when playing—usually without using headphones. The room isn't boomy nor is it ringy or metallic sounding. In short, music sounds great in the Telearte studio; I thought so and the musicians coming in to record there thought so too.

Half the studio is deadened with Sonex acoustic foam and carpeting while the other end is live with hardwood floors, walls and ceilings. The live end of the studio utilizes polycylindrical diffusers on the wall opposite the dead end, and large, oval-domed diffusers (similar to sky-lights) on the ceiling.

WE BUILD 'EM STRONG

Chips specified particularly rugged construction plan details which, I'm afraid, has permanently "endeared" him to the Venezuelan carpenters. For example, wall studs were spaced at four inches with solid, real wood paneling (the studio's acoustic boundary) *screwed* to the studs each four inches both vertically and horizontally. What the exhausted carpenters actually did was to stud each six inches and apply wood screws each six inches horizontally and vertically. Even so, that's an impressive number of wood screws when you consider the massive surface area of both the studio and the control room. The payoff: When you knock on one of the walls you get no "thump" or "boom" sounds coming back (the wall does not act like a huge drum head). The studio and control room are separated by two 15 mm thick panes of glass. Only one pane of this heavy glass is used, however, between the isolation rooms and the studio (and



Producer's-eye view looking over the keyboards to the drum and bass area.

between the isolation rooms themselves). If there was one thing that the engineers at Caracas might be tempted to change, doubling up the iso room glass application might be it.

Aside from the iso rooms being not very isolated, all of the precautions and design effort worked: you can make as much noise as you like in the studio and absolutely not a sound can be heard in the control room.

The "wing" iso booths are mainly deadened by Sonex foam. The central iso booth (which is quite large) takes a different approach—removable Sonex panels over hard wood walls can make it live, dead or a little of both (?). The Sonex panels are mounted by velcro fasteners and are easily "popped" off the walls giving either a live, partly live, (partly foamed), or dead booth. This is going to come in very handy for the engineering staff, considering that they might want to record drums, vibes, guitars, percussion or vocals in the large central booth.

Once the control room was ready, Don McLaughlin of Sphere arrived to complete the installation of his Eclipse "C" console. The Eclipse fits well, has a lot of useful functions, is easy to use, looks good, sounds crystal clear and the music comes through hard as nails even with the extremely telling LEDE Time-Aligned, room speaker combination. The console went in, lit up, worked fine and Don went home.

The Sphere board is very well made both physically, logically

The LEDE control room at Telearte with time-aligned monitors, Sphere console, and Ampex ATR-124 multi-track recorder.

and electronically. Personally, I would prefer a separate monitor/cue section (not everything "in-line"), especially when a team of engineers splits up session chores to achieve speed. Yet even with the Eclipse's in-line monitor design, a sort of team effort was possible with a minimum of reaching over one another: the engineer handling the I/O, levels and equalization and the assistant handling the various cue mixes (while wearing headphones) and helping with the monitor mix (while not). In the recording technique I teach, the monitor mix is always recorded on open tracks of the multi-track machine ("poor man's automation"), and so the quality and level acceptability of the monitor (stereo) bus is quite important. Therefore, the assistant (headphones off for the moment) sometimes has to back down all the individual monitor levels a little (to bring the stereo bus into acceptable level limits) while counter-adjusting the monitor master or stereo master to maintain a constant monitor level in the speaker system.

Sphere consoles utilize hand-made, discrete amplifier modules which are used throughout the various stages of the signal path. These little amps sound very, very good. If you purposely make them distort, they also sound very good—a little like old-time tube distortion. We patched audio 24 times through the console to A/B the first time through vs 24 times through. The difference in sound quality was *remarkably* small.

Sphere's little "building block" ampifiers have a feature which really comes in handy during times of trouble. Each little amplifier has two tiny lamps mounted on it, one or both of which will light up if the amplifier should fail. To locate which modules have malfunctioned, you have only to open up the console's back and replace the amplifier modules which have lit up! None failed during my five week stay, but if one had, it would have taken only about a minute to find and replace it with a spare. What a marvelous idea. Another marvelous idea was mounting all the mother-boards on hinges so that a maintenance person can just swing them out and over for checking into the board's underside.

Easy maintenance is very important in a country with less than reliable AC power and lengthy send-out maintenance cycle times (especially if the send-out is back to "the States"). One evening, while we were doing a practice tracking session with the help of a very energetic Salsa band, we had four- count, four—total power failures. One of these occurred during a full-tilt, flat-out rewind of two inch tape on the ATR-124. To all of you who know how fast an ATR-124 can go during shuttle (the speed is settable by a trimpot—we had it quite fast), you can imagine the unusual horror of being suddenly in the dark with the worrisome sounds of decelerating tape shuttling somewhere near the wall. When the lights came on again, we found that everything was O.K..It is to the credit of Sphere, Ampex and Urei that even multiple Caracas power failures do not result in blown speakers, destroyed tape or downed components!

A SIDE TRIP FOR SHERMAN

Before continuing with the recording article proper, I would like to digress a little. Peter Bottome, knowing that I was an

The PB-Y Flying Boat

airplane enthusiast (I'm a commercial pilot), decided to take me on a mini-expedition to a far distant island off the eastern coast of Venezuela. We took the ride to the island in Peter's PB-Y flying boat. This is really a *large* airplane; it moves quite slowly (about 100 MPH), but has a long range. Its two engines require 30 gallons of oil *each*. It is larger than a large school bus inside and had been outfitted as a camper similar to Cousteau's PB-Y (before its sad crash). Well, I flew this beautiful dinosaur from the past and I can tell you—it's quite a bundle of joy while sitting at the wheel. I flew for about an hour and then Peter took over and landed us on the water at the most beautiful little island you can imagine. Just before landing, the PB-Y's wing tips are folded down into little stabilizer boats so that a wing tip won't go under while the plane (boat, really) is at anchor. Out of the plane's many storage areas comes a Zodiac rubber raft (a la Cousteau). Up onto the enormous wing it goes with a scuba bottle of air to be inflated. Onto the raft (now dropped from the wing into the water) goes the 25 HP outboard motor and away we went—I to sun and swim on the beach (it was my day off), Peter to search for a wind recorder device he had planted on the island one year ago (he didn't get it back—it had been stolen). This is one day I will remember forever. Thanks again Peter!

Dead End showing Sonex and carpet application, instrument areas, and the three vocal booths.

MEANWHILE BACK AT THE STUDIO...

Now—as though it had fallen from audio heaven right onto Teleart's vacant lot—we have a truly beautiful and ultra-modern studio of the most advanced acoustic and electric design equipped with the best of everything. Truly a recording facility which has gone beyond the expectations of most of Venezuela's music professionals.

In the studio, session operation was greatly streamlined by the development of a semi-permanent setup using the studio's own drums, bass, keyboards, electric guitars and amplifiers. The studio's standard setup included chairs, music stands, headphones, mikes and direct boxes. Mike and direct inputs were provided for all the electric instruments. Drum mike application was developed so that either a single or dual overhead recording style or, "a one mike per drum" style could be used with minimum console and microphone placement changeover. A Yamaha 8/2 submixer was used to combine sounds from the three electric keyboards into the keyboardist's instrument amplifier. The keyboard instrument amplifier was only used for the player's acoustic monitoring; each keyboard was recorded using its own direct box. PZM® mikes were used for distant drum overheads, stereo room pickup, vocal group pickup and room talkback. The three vocal booths were permanently miked and headphoned for instant use on the spur of the moment. The "microphone list" (one of the session documents I retained) goes like this:

Console Input	Instrument	Mic
1	Kick drum	441
2	Snare	SM57
3	High-hat	451
4	Tom	SM57
5	Tom	SM57
6	Tom	SM57
7	Overhead	451
8	Overhead	451
9	Bass	HNE Direct Box
10	Bass	D12
11	Electric guitar 1	HNE Direct Box
12	Electric guitar 1	441
13	Electric guitar 2	HNE Direct Box
14	Electric guitar 2	441
15	Keyboards OBX	HNE Direct Box
16	Keyboards Chroma	HNE Direct Box
17	Keyboards EKB	HNE Direct Box
18	Keyboards Yamaha	451
19	Keyboards	451
20	Vocal 1	U87
21	Vocal 2	U87
22	Vocal 3	U89
23	Vocal 4	U87
24	Studio talkback	PZM
25	Extra 1	414
26	Extra 2	414
27	Drum Machine	HNE Direct Box
28	Open	
29	Open	
30	Open	
31	Open	
32	Open	

Thus, no matter how many musicians walked in to record (in contrast to the usually unreliable session information gathered by the "traffic" office), the studio is ready to go. Towards these ends, a basic, minimal equalization scheme was slowly worked out and kept in his notes by each engineer which could be "dialed in" as part of normalizing the console for tracking when the in-house instruments were to be recorded. For privately owned instruments brought into the studio, time was taken, of course, to get "custom" levels and equalization from "scratch."

Upon walking into a session, the console was always found pre-marked with *all* the inputs of the semi-permanent setup. An advantage of this system for Telearte was that all mikes, directs and headphones were already tested and known to be working and noise and distortion free. The artist could come to the studio, sit down at one of the instrument areas, and begin playing—the overdub could be going to tape in mere seconds.

Since the studio only uses Ampex tape (456), the machines are always aligned and ready to go using the studio's stock levels, azimuth and equalization (as provided by Magnetic Reference Laboratory tapes).

The recording sessions are a bit more relaxed in Caracas than they are in L.A. The instrumentalists rarely bring their own amplifiers and some don't even bring their own instruments. Telearte Studios has, therefore, a nice selection of instrument amplifiers, an electric bass and an electric guitar—all by Yamaha. In addition to the Yamaha acoustic piano, there is a Rhodes electric piano, a Rhodes Chroma, an Oberheim OBX, and an Oberheim DSX and DMX (sequencer and electric drummer). If the session players fail to show up on time (a distinct likelihood considering the enormous number of cars and the inadequate number of streets), you can make quite a lot of music singlehandedly with all the computer-controlled synthesizer equipment at hand. ■

SIDNEY L. SILVER

Electronic Devices With the Power of Speech

The use of synthesized speech is attracting widespread attention in the broadcast industry. Here, author Silver outlines some of the applications and problems of this emerging technology.

PICTURE THIS SCENE: You are an on-call broadcast engineer and you must check the performance of a remote, and unattended transmitter. To do so, you call a special number using a touch-tone signalling system as a data-entry device. The coded tones generated at the terminal are carried over the telephone line to interrogate a microcomputer interfaced to the transmitter's automatic control system. The microcomputer decodes the touch-tone signals and issues a request to a voice response synthesizer to assemble a reply. This is accomplished by accessing the proper sequence of words stored in ROM (Read-Only Memory) to create the appropriate message. The synthetic verbal reply is then returned to you, communicating the present equipment status and any other pertinent information.

'DON'T CALL IT; IT'LL CALL YOU'

Now imagine the scene changes: A maintenance problem occurs at the transmitter, and requires immediate attention. When the fault condition was recognized by the transmitter control system, the unit first tried to make its own corrective adjustments by following stored instructions. But the problem is too complex for the unit to handle on its own, so the microcomputer initiates a series of telephone calls to try to locate you. Here an automatic dialer reports any of, say, ten alarm conditions to a list of phone numbers. The number of any alarm triggered by a specific malfunction in the transmitter is beeped out and a synthetic voice states the problem in precise terms over the telephone line. Simultaneously, the same information is fed to the house monitors in the studios. Thus, if a service call must be made, the problem has been isolated in advance with minimal downtime.

The use of synthesized speech described above represents only one of many applications that are generating considerable interest in the broadcast industry. Another application which requires less support involves making synthetic verbal announcements in information-retrieval systems. For example, a voice identification can be used to announce the number and title of master tapes on file in the production room. Speech synthesizers can also be interfaced with weather computers to provide a telephone caller with direct access to current weather information reported on a continuous basis. One can envision a nationwide system whereby the caller dials up the desired area code, using touch-tone entry, and receives via synthetic speech an up-to-date computer-generated weather report from any region in the country.

In conjunction with digital speech synthesis, work is being done on the development of closely related speech recognition techniques. In contrast with speech synthesizers which generate a pre-selected arrangement of words, speech recognition schemes convert verbal commands by humans into digital character strings which actuate certain control functions. However, this technology lags far behind speech synthesis and will not be discussed in this article.

THE PROBLEM OF STORAGE SPACE

In considering the application of digital processing for speech synthesizers, it is important to consider the need to conserve memory space in voice response systems. Basically, to produce synthetic speech, we require an efficient means of storing a vocabulary of words, phrases, and sentences, together with a suitable means of retrieving selected elements of the vocabulary in a prescribed sequence. Perhaps the simplest and most direct way of doing this is to convert original spoken words into digital codes with an analog-to-digital converter, and store them in a ROM. Later, the system will decode the contents of the ROM, reconstruct the codes in their proper order, and transform them back to speech with a digital-to-analog converter. For example, using linear PCM, each amplitude of the speech signal can be represented by a different code, with the basic sampling rate determined by the desired speech intelligibility. Here the main disadvantage is the enormously high data rate required to store the speech sounds. Let us assume, for example, that acceptable speech quality can be obtained with a signal bandwidth of 4 kHz, requiring a minimum sampling rate of at least 8 kHz. If the encoded samples are represented by 8 bits of data, the effective data rate would be 64 kbits/sec. At this high rate, even a small vocabulary would demand large storage capacity, adding considerable cost to the system.

Consider a voice response system with a vocabulary of 100 words whose average duration is one second. At a rate of 64 kbits/sec, the digital storage requirements for 100 seconds of speech material would be 6.4 Mbits/sec. Putting this amount of

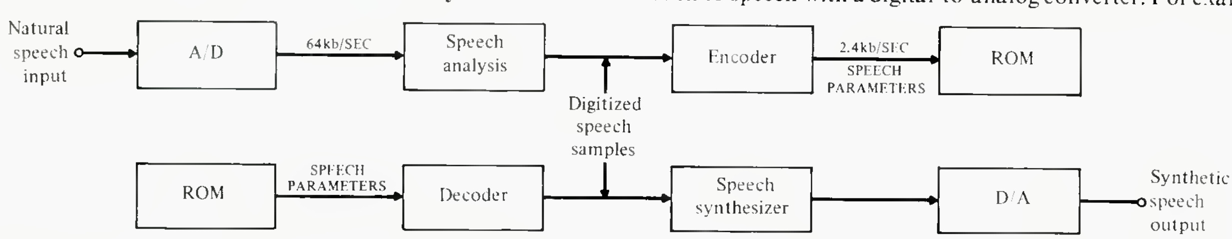


Figure 1. The speech processor is composed of two subsystems; Analysis and synthesis. Data is compressed for storage during analysis, and expanded upon retrieval during synthesis.

Sidney Silver is on the supervisory staff of the Telecommunications Section of the United Nations, where he is in charge of sound and recording.

storage space into perspective, it would take only a few seconds of speech encoded with linear PCM to use up all the storage space in many types of available memories. It is possible, of course, to reduce the data rate by using various forms of differential coding, e.g., adaptive differential PCM, continuously variable-slope delta modulation, etc. These schemes take advantage of the statistical properties of speech by removing some of the excess, or redundant data, from the speech signal, bringing data rates down to about 24 kbits/sec. But even so, storage requirements would still be too large for even a small vocabulary. Clearly, these methods are impractical unless further compression techniques are applied to the digitized waveform.

MODELLING THE VOCAL TRACT

There are other types of speech coding available which can be used to digitize speech at a much lower rate than is possible with waveform digitization methods. One example is the analysis-synthesis system shown in FIGURE 1. Since speech signal patterns are highly redundant and somewhat predictable, no attempt is made here to actually preserve the original speech waveform. Instead, the voice is encoded using digital parameters developed from a careful analysis of the analog speech to simulate, or model, the human vocal tract. An analysis is performed by breaking the speech patterns down into certain parameters, i.e., pitch and energy components, each of which are stored in digital memory in a compressed state. The result is that significant bandwidth reduction can be realized (on the order of 1200 to 2400 bits-per-sec) without sacrificing voice quality. During the synthesizing procedure, the speech signal is retrieved from memory and reassembled using information derived from the stored parameters.

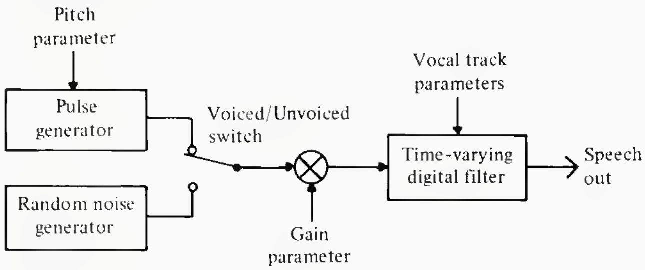


Figure 2. Block diagram of a simplified model for speech synthesis.

A valid approach on which to base the processing of speech signals is to represent these signals in terms of a simplified model, such as depicted in FIGURE 2. This is a linear system whose output has the desired speech-like properties when controlled by a set of slowly-varying parameters that closely resemble actual speech production. To produce a speech-like signal, a time-varying digital filter is used to approximate the spectral characteristics of the human vocal tract. Input to the filter is provided by either of two modes of excitation. For voiced speech, the filter is driven by an impulse-train generator that creates a near-periodic pulse stream. The spacing between pulses corresponds to the fundamental period of the speech waveform. For unvoiced speech, all that is required is a random noise generator that produces a flat spectrum noise. Now, by switching between the voiced and unvoiced generators, we can model the changing modes of excitation.

Since the human vocal tract changes its shape rather slowly in continuous speech, it may be assumed, for practical purposes, that the characteristics of the speech waveform are nearly constant over a short time interval, say, about 10 to 30 msec. Based on this assumption, the digital filter may be characterized during each interval by a set of weighting factors, or filter coefficients, which emulate the slowly-changing vocal tract parameters. Thus, from the point of view of speech analysis, techniques can be worked out for estimating the parameters of the model derived from natural speech sounds. For speech synthesis, on the other hand, we can use the model to create a new waveform by controlling these parameters. Ideally, if the model is sufficiently accurate, and the parameters precisely determined, the resulting output of the system should be indistinguishable from natural human speech.

LINEAR PREDICTIVE CODING

One of the most powerful techniques for extracting human speech parameters, and restructuring synthetic speech from these parameters, is the method known as Linear Predictive Coding (LPC). The importance of LPC lies in its ability to provide extremely accurate estimates of speech parameters, as well as producing synthetic speech with greater naturalness than many other types of analysis/synthesis systems. In the analysis procedure, the basic idea is to record and compress a human voice and develop digital data that will, on command, drive a speech synthesizer. To do this, the present value of a speech sample is approximated as a linear combination of past speech samples. By minimizing the differences (over a finite period) between the actual speech samples and the linearly predicted ones, the appropriate filter coefficients are determined. These coefficients can then be used to drive the digital filter located in the synthesizer section.

Besides estimating the filter coefficients, the analysis procedure also decides whether the speech signals are voiced or unvoiced. If a voiced signal is detected, the analysis computes the fundamental frequency, or pitch period, of the signal. A calculation of the energy level is made to determine the gain needed by the synthesizer to reproduce a source with the same intensity as the original speech signal. All of the parameters described above must be updated regularly at a faster rate than it takes the human voice to cause a significant change in the vocal tract configuration. It has been demonstrated that if the speech parameters are adjusted about 50 times per second, a sufficiently smooth transition can be achieved between one speech segment, or data frame, and the next. This is equivalent to compressing the bandwidth of the digital data within each frame every 20 msec.

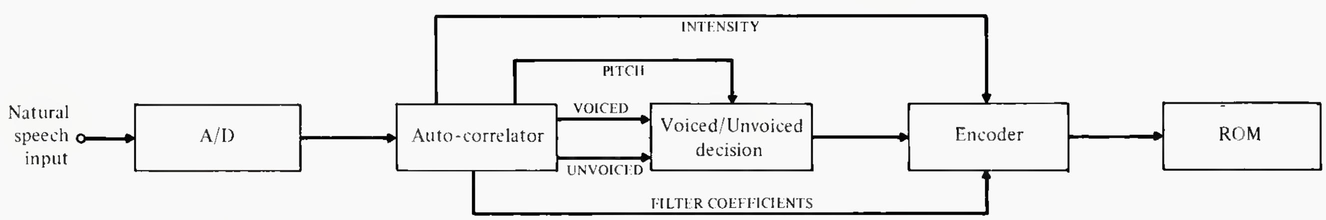


Figure 3. A Linear Predictive analysis system extracts parameters derived from the speech waveform and encodes them at a low bit rate.

To accomplish this, an autocorrelation function is applied to the analysis system (FIGURE 3) to extract detailed information about the spectral content of the input speech signals. The autocorrelator computes two sets of coefficients for each 20 msec frame, one set for tracking the pitch of the speech signal, and the other for providing the weighting factors for the tenth-1 order digital filter in the synthesizer. For each data frame, the filter coefficients are calculated by obtaining the solution of a 10-dimensional set of simultaneous equations. The solutions of these equations, representing the 10 coefficient values, are then permanently stored in a ROM along with the two sets of values describing the energy and pitch characteristics of the speech samples.

Altogether, a total of 12 parameters are needed for each 20 msec frame period. If, for example, each parameter is given a 4-bit code, then a minimum rate of 2.4 kb/sec is required to satisfy this condition (i.e., 12 x 4/20 x 10^3). Within one second there are 50 frames, so that at a rate of 2.4 kb/sec, each frame holds 48 bits of information (2400/50). In practice, however, it has been found that not all speech requires a 48-bit representation. This fact can be used to minimize the amount of memory space necessary to store phrases and sentences. As an example, during the production of unvoiced speech the speech processor in the digital filter circuitry. Using this approach, arithmetic calculations can be performed in stages, so that after the data is processed in a given stage, it is passed on to the next stage while a new multiplication and accumulation is begun in a previous stage. Thus, more than one arithmetic calculation can take place simultaneously, with each one in a different stage of completion. Modelling the vocal tract in this way adds the quality that makes synthetic speech sound natural.

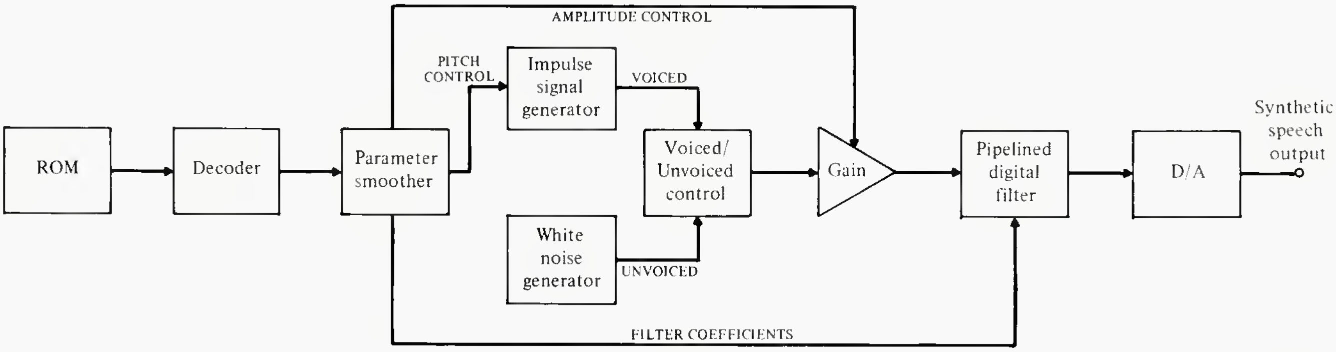


Figure 4. The synthesizer uses linear predictive coding to control voice parameters and reassemble a synthetic version of the original speech waveform.

samples may be represented by as few as 4 filter coefficients, so that these samples may be encoded by approximately 28 bits rather than the usual 48 bits. Also, it has been found in many cases that the spectral characteristics of voiced sounds, and hence, the filter coefficients, may not change significantly for several consecutive frames, say, over a 100 msec period. When this happens, it is unnecessary to encode the same 48 bits in each 20 msec frame. Instead, for each 100 msec voiced sound, the first frame can be described by 48 bits, but the following four frames are repeated, rather than encoded as an entirely new bit pattern. Since a repeat frame is designated as 10 bits in length, a saving of 38 bits per frame can be realized for a voiced sound. The variable-length bit pattern thus created reduces the effective data rate to less than 1.2 kb/sec without degrading speech quality. This represents more than a 50-to-1 reduction in storage space over direct waveform encoding, such as exemplified by linear PCM.

LINEAR PREDICTIVE CODING—SYNTHESIS

Restoring the digital data to its original analog form, or speech synthesis, is essentially the reverse process of speech analysis. Initially, the data stream is unpacked from the memory and decoded into appropriate control parameters, then the sequence of bits is translated into a linear approximation of the original speech waveform. As shown in FIGURE 4, the pitch and energy data, and the 10 filter coefficients, are separated in the synthesizer before being applied as excitation sources. The pitch data, for example, consisting of a bit stream spaced one pitch period apart, is used to excite the impulse signal generator. If the pitch data is zero, that is, unvoiced, the excitation function is obtained from the white-noise generator. The selection between the two excitation generators is made by the voiced/unvoiced control switch. In more sophisticated designs, the operation of both generators can be combined when necessary to generate certain voiced fricative sounds. To provide a gain control, the overall amplitude of the excitation is derived from the energy parameter.

In this process, each data frame contains 160 speech samples. We obtain this figure by dividing the 20 msec frame by the 125 $u$sec sampling period. The pulse stream is then weighted by the tenth-order digital filter (usually employing a lattice structure), which performs about 1600 multiplications and accumulations on the 160 samples per 20 msec frame. However, because of the large number of arithmetic operations that must be carried out, and the limited time allotted to perform this function, it is desirable to enhance the filter speed characteristics and permit a more efficient operation. One way to produce a high-speed operation is to incorporate a pipeline

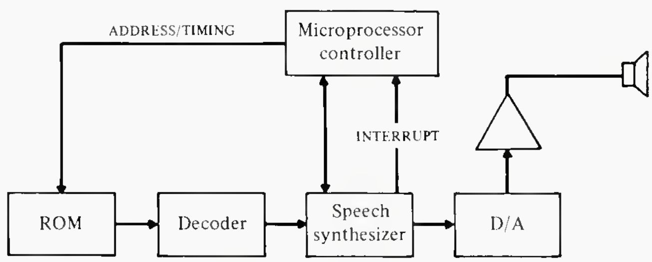


Figure 5. A simplified voice synthesizer provides speech from a fixed vocabulary stored in an external Read-Only Memory.

In order to remove abrupt changes between digital samples, such as those produced by the transition from a voiced data frame to an unvoiced one, a parameter smoothing circuit is generally used. The smoothing circuit slowly attenuates the amplitude parameter at the end of a voiced frame and, at the same time, gradually increases the amplitude at the start of the succeeding frame, from zero to its defined value. By this means, the end points of each frame are initially kept low in amplitude to even out the contours of the synthesized speech waveform. This can be accomplished by interpolation, whereby a small portion of the difference between two successive digital samples is repeatedly added to the first sample. The resultant smoothing action helps prevent pops and clicks and other transient impulses from passing through the synthesizer.

INTERFACING

There are many configurations that can be used to interface a speech synthesizer with a control function. FIGURE 5 shows one way a microprocessor can control the operation of a synthesizer—by instructing it which words to say and when to say them. More specifically, the microprocessor controller determines which elements of speech are to be spoken and where they are located in the ROM. Then it reads the data from the ROM and delivers this information to the synthesizer in the required sequence. In order to form phrases and sentences from individual words, the controller addresses data from the ROM and issues a talk signal to the synthesizer. The synthesizer then links the appropriate words at a rate depending on the data processing time. At the output of the system, analog speech is delivered to an external amplifier which, in turn drives a loudspeaker. When the controller becomes inactive, that is, standing by to load a new word address, an interrupt signal is fed to the controller signifying that the synthesizer has stopped talking. To facilitate this sequence of operations, a real-time clock, or event timer, is provided in the controller circuitry. ■

New Products

PRO POWER AMPS

• The CM 915 professional power amplifier delivers 150 watts per channel into 8 ohms, and 225 watts into 4 ohms. Input levels are controlled by eleven position stepped attenuators; output power levels are indicated by use of ten color-coded LEDs. Front-mounted professional circuit breakers offer speaker protection, and a pair of amber LEDs indicate possible overtemp conditions before sonic degradation or amplifier damage can take place. The unit can be used in the bridged mono mode.
Mfr: CM Labs

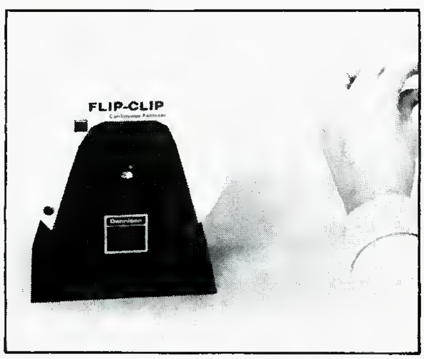
Circle No. 31 on Reader Service Card

AUDIO TAPE CLEANING FABRIC FOR TAPE DUPLICATORS

• Available in ½-, 1-, and 1½-in. widths on 100 foot or 200 foot rolls, Tapemaker audio tape cleaning fabric is designed to remove debris and dirt from all types of magnetic tape. (Custom widths and lengths are also available.) Audio tape cleaning fabric is ideal for use on Electro-Sound, Ampex and Gauss duplicating equipment as well as for use on slitters.
Mfr: Tapemaker

Circle No. 32 on Reader Service Card

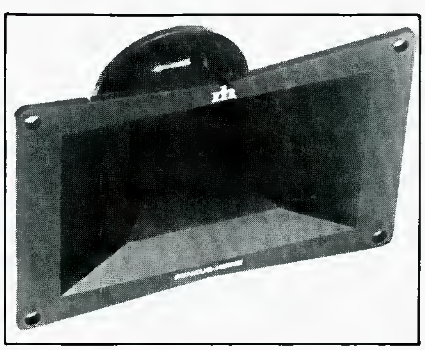
FASTENERS



• Available in 500 or 1,000 clip reels held in a bench-stand dispenser, the new Flip-Clip™ fasteners can be positioned, flipped close and snapped off the continuous strip in one easy motion. No tools are needed and there is no time wasted in picking up and orienting individual pieces during assembly operations. The Flip-Clip fasteners feed directly from the reel onto the item to be bundled, secured or tied. Flip-Clip fasteners can be repositioned, removed or reused easily. Molded from type 6/6 Nylon in natural, black or special colors, Flip-Clip fasteners have a loop tensile strength of nine pounds and will bundle diameters up to one-half inch with a simple twist.
Mfr: Dennison Manufacturing Corp.

Circle No. 33 on Reader Service Card

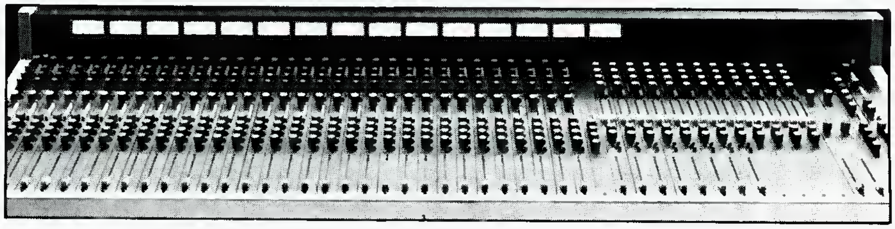
CONSTANT BEAMWIDTH HORN



• The Model CBH 1600 1-in. throat horn has a cutoff frequency of 1600 Hz and is designed for constant coverage in the horizontal and vertical planes. A fast flare rate integrated with a short path length reduces 2nd harmonic distortion and allows mounting in the low frequency driver's acoustic plane for a time coherent source. The CBH 1600 is designed for near or far field monitoring applications. The CBH 1600 horn mounts easily into any speaker enclosure. The front flange is large enough to permit front servicing of most 1-in. drivers.
Mfr: Renkus-Heinz

Circle No. 34 on Reader Service Card

RECORDING CONSOLES



• The R-16 and R-24 Series are modular plug-in design consoles intended for 8- and 16-track recording. With the addition of R-24 expander modules, 24-track recording can be accomplished as well. The consoles employ separate input and group/output modules. R-16 and R-24 consoles have individual channel cue/effects feeds and two bus cue/effects feed controls to accommodate musicians with varying cue requirements. Standard input module features include 3-band fully sweepable equalization, switchable phantom power, phase switching, HPF, left-right and odd-even bus assignment, four effects/cue buses, and Audioarts Engineering M-104 conductive plastic linear faders. Console systems include LED clip overload indicators throughout the signal path, separate effects send and return modules, master left and right output modules, and a CS-8 control module housing full studio, control room and headphone monitor, talkback, slate, oscillator, and solo functions.
Mfr: Audioarts Engineering
Price: between $10,000 and $17,000
Circle No. 35 on Reader Service Card

ANTI-CORROSION CAPSULES

• Audiotechniques has recently been appointed distributors for Zerust Anti-Corrosion capsules. Zerust is a new anti-corrosive device that emits a continuous vapor which forms a protective thin film on metal surfaces. The VC-1 capsule will protect a one cubic-foot area and has a life span of one year. Other models are available with up to six cubic-foot protection and two year life. Zerust works best within enclosures where air exchanges with the outside atmosphere is limited. Testing has proven that the vapor will not increase contact resistance in relays and switches.
Mfr: Audiotechniques Inc.
Price: $92.50 a case
Circle No. 37 on Reader Service Card

FLOPPY DISK HEAD CLEANER

• The Model CMP-145 Floppy Disk Head Cleaning System consists of 8 pure polyester cleaning diskettes, a 1.5 oz. Freon-based cleaning spray, reusable diskette jacket, software programming and complete instructions. The cleaning system removes dust, dirt, smoke particles and oxide particles from floppy disk read/write heads. It provides longer life for heads and media diskettes employed in single or double-sided drives, and prevents read/write errors, data loss, downtime, service charges and damage to read/write heads and media. Included in the CMP-145 is a set of programs written in BASIC language for Apple II, TRS-80 III, IBM-PC and others. The system provides 4 fresh cleanings per diskette by stepping the read/write head to 4 fresh locations per diskette. It also provides non-stop, 30-second drive action.
Mfr: Nortronics
Circle No. 36 on Reader Service Card

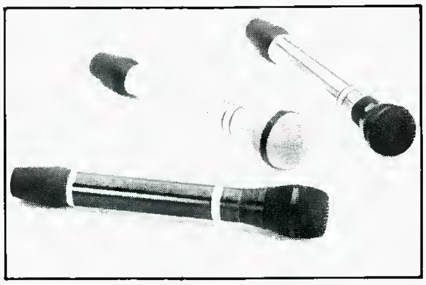
MODULAR AUDIO PROCESSING SYSTEM

• MicMix has recently announced production of the MC-Series Modular Audio Processing System. Currently included in the series is a 5¼-in. rack-mount Card-Frame, as well as two signal processing devices, the MC-101 and the MC-201. The MC-F Card-Frame will accept up to five individual modules, along with a non-powered space for storing an extra module. The MC-F is powered by a low-noise, toroidal transformer that allows all powered modules to 600 ohms. The MC-F is compatible with the dbx® 900 series, as are all the MC-Series modules. The MC-101 is a single channel of Dynaflex noise reduction that is said to provide up to 30 dB or more of noise reduction without encoding or decoding. The MC-101 includes a Threshold control for noise reduction, a hard-wired In/Out switch, and a switch to set the appropriate signal reference level from −10, 0, +4, or +8 dB, allowing interface with consumer, sound reinforcement, recording studio, or broadcast equipment. The Master-Room MC-201 allows the user to vary the decay time of virtually any reverb device, along with providing up to 30 dB of noise reduction. This is accomplished by utilizing patent pending downward expansion circuitry. The MC-201 includes a Decay control, a hard-wired In/Out switch, along with the adjustable reference level. The MC-201 operates before the reverb return function of any console, and will allow the user to shorten the decay time by up to 75 percent. It can be used to vary the decay time of any plate (without damping), a live chamber, or any spring system without altering the tonal characteristics of the reverb device.

Mfr: MicMix

Circle No. 38 on Reader Service Card

WIRELESS HANDHELD MICS



• Swintek now offers a choice of wireless handheld microphones and finishes. Buyers can choose the Beyer M500 ribbon capsule, the Shure SM57, SM58, and SM78 dynamic capsules, or the Shure SM85 electret condenser capsule. Finishes can be ordered in chrome, black chrome, or gold. All mics include Swintek's "dB-S" companding system, which delivers over 80 dB dynamic range. Swintek uses the VHF/UHF high band to avoid interference from CBs and business radios, and to prevent the mic from interfering with video equipment. Narrow band FM transmission is also used, which not only avoids interference from nearby radio and television stations, but enables more mics to be used on adjacent frequencies without interference. The mic transmitters are built with glass-epoxy circuit boards, joined by gold-plated connectors and shock-mounted inside metal cases. Battery life is typically 10 hours using a standard 9V alkaline battery or Swintek's rechargeable THR-B Ni-Cad, and range is typically 1,000 feet (using a Swintek wireless mic receiver). A power on-off switch and a modulation level control are provided.

Mfr: Swintek Enterprises, Inc.

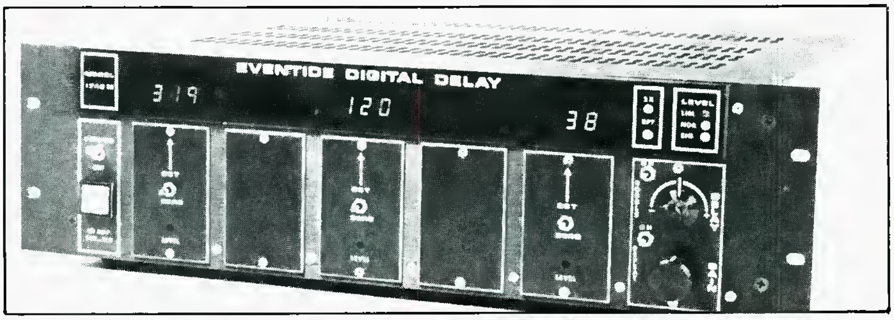
Circle No. 40 on Reader Service Card

MODULAR DIGITAL DELAY

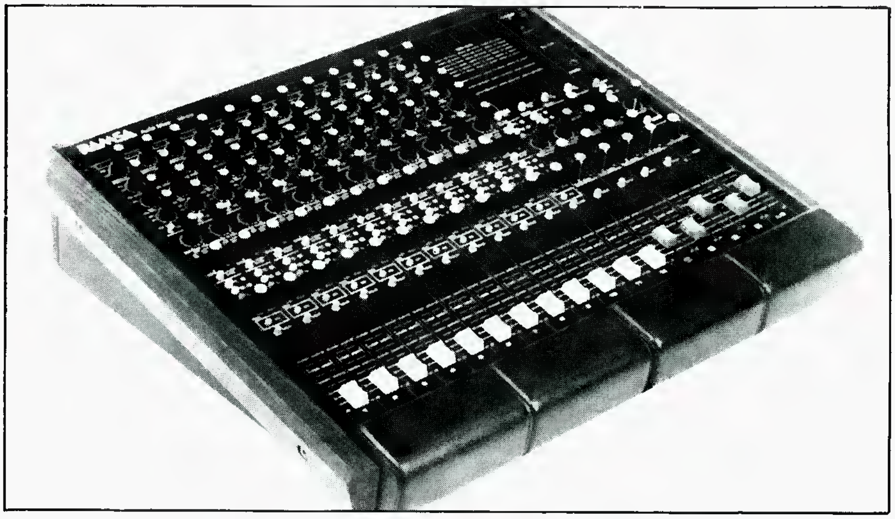
• A new version of Eventide's 1745M modular digital delay is now available, offering maximum delay lengths eight times longer than previous models. Delays of up to 2.5 seconds (5 seconds in DOUBLE mode) are now available to each output (up to 5 output modules can be installed in the mainframe). Frequency response remains 50-15 kHz ±1 dB at all delays up to the new 2.5 second maximum. The increase in performance—made possible by the new generation of RAM chips—is offered at no increase in price. To accommodate the 1745M units working in the field, Eventide is making a retrofit kit available.

Mfr: Eventide Clockworks

Circle No. 39 on Reader Service Card



RECORDING CONSOLE

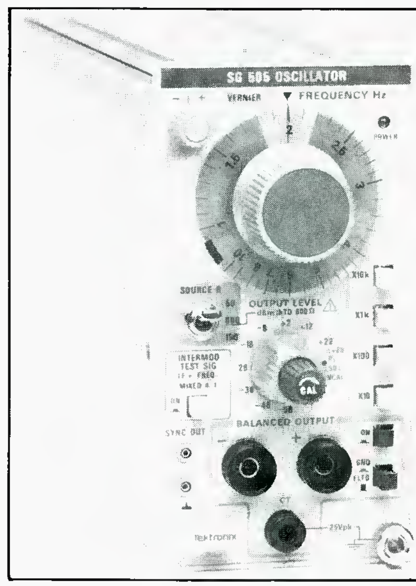


• The model WR-8112 sound reinforcement and recording console is designed with 12 mic and line inputs and provides the versatility of trim, monitor/effects send, solo controls, stereo effects returns and cue send outputs. The outputs include 4 Group, 2 Master, and 1 Mono Master. In addition, a 12-point LED meter can measure any signal that travels through the console. A flexible set of controls directs both sound reinforcement and recording functions, including a 3-band equalization section on each input, covering high, mid-range and low frequencies. The high and low knobs are equipped with a 2-position frequency selector for versatility. Control over such frequencies as vocals and brass are provided through a sweepable, peak-dip-mid-range knob, covering a more varied range than the fixed type. Direct outputs are provided on all inputs.
Mfr: Panasonic
Price: $2495.00
Circle No. 41 on Reader Service Card

PHASE CHECK SYSTEM

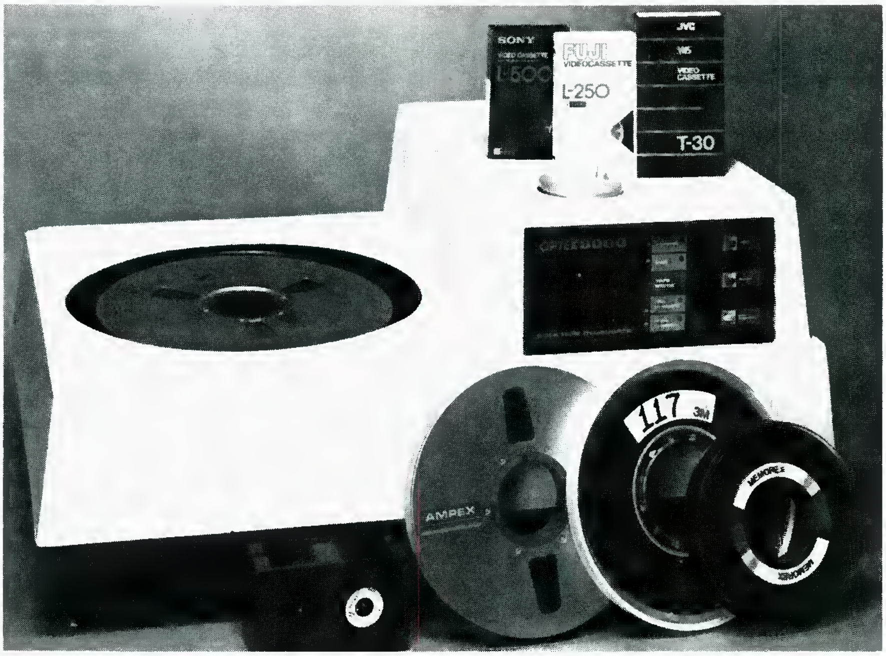
• The AR130S and AR130D two-unit test system is designed to check the phase integrity of most two-wire electrical equipment such as connecting cables, electronic equipment, microphones and loudspeakers. The AR130S is the symmetrically encoded tone source, which is connected to the output being tested. Microphones can be directly connected to the detector input for microphone and loudspeaker checking. The Phase Check system feature features an encoded test signal that enables reliable testing of loudspeakers, including high compression drivers; four presettable tone frequencies; variable output level and battery "low" indicator. No direct connection is required between source and detector, other than the signal under test, which may be a single item or a complete system. All circuitry except batteries are housed inside a steel box for protection.
Mfr: Brooke Siren Systems
Circle No. 42 on Reader Service Card

OSCILLATOR



• The SG 505 Options 2 oscillator provides a fully balanced, fully floating output with a maximum calibrated amplitude of +22 dBm from 600 ohms into 600 ohms and +28 dBm from 50 ohms into 600 ohms. Into 150 ohms, more than +30 dBm can be achieved. More than 100 dB attenuation of the output level can be achieved with the built-in step and variable attenuators. The key performance features of the standard SG 505 have been retained in the Option 2—0.0008 percent maximum THD (typically 0.0003 percent) from 20 Hz to 20 kHz and ±0.1 dB flatness from 10 Hz to 20 kHz. When used with the AA 501 Distortion Analyzer, the SG 505 Option 2 can generate either the SMPTE or DIN Intermodulation Distortion (IMD) test signals. Selection of the IMD signal switches on an internal low frequency oscillator whose output is combined 4:1 amplitude ratio with the selected output frequency. The AA 501 Distortion Analyzer will then automatically measure the intermodulation distortion. Packaged as a plug-in for the TM 500 family of modular test and measurement instruments, the SG 505 Option 2 can be combined with the user's choice of over 35 other plug-in instruments.
Mfr: Tektronix, Inc.
Price: $1360.00
Circle No. 43 on Reader Service Card

HIGH-POWER DEGAUSSER



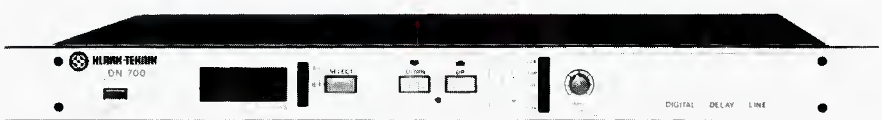
• The Christie MAXERASE-16 Degausser erases computer tapes and video tapes in 30 seconds on a continuous production basis. 2-inch master tapes can also be erased, but must be passed through twice (once on each side). The MAXERASE-16 achieves degaussing up to −95 dB (below recorded level) on all computer tapes. To obtain the high-power erasure, Christie uses a patented core assembly which locates one core just under the reel tray in close proximity to the bottom edge of the tape. The top core can be raised or lowered manually to provide a close working tolerance of the core to the top edge of the tape. The two cores are magnetically coupled, and are specially shaped to minimize horizontal flux patterns which heat up the reel but do not effectively degauss the tape. The tape is automatically moved through the lines of flux in a rotary motion in order to avoid thumps, spoking or other recorded-in residue left on the tape. The MAXERASE is built with an enhanced duty cycle for long-term operation without overheating, and has switch-selectable high and low power settings for degaussing versatility. High power operation will begin to heat up the coils to a "temp alarm" condition after an hour of heavy usage. Low power operation is virtually continuous and suitable for most tape erasing applications except applications above one inch in height, such as high coercivity two-inch tape.
Mfr: Christie Electric Corp.
Price: $6500.00 for the base unit
Circle No. 44 on Reader Service Card

TWO-WAY SYSTEM

• The model 4691 compact two-way system incorporates the recently developed 2370 flat-front bi-radial horn, a 2425J titanium-diaphragm high frequency compression driver, and an E140 15-inch woofer. The 2370 bi-radial horn provides improved on- and off-axis frequency response in the horizontal plane, with a 90 degree horizontal by 40 degree vertical nominal coverage pattern to beyond 16 kHz. Use of JBL's 2425J compression driver results in expanded frequency response, as well as high efficiency and power handling. A 1.5 kHz high pass network blends the low and high frequencies, while switchable biamplification inputs are conveniently featured on a rear terminal panel. The 4691 may be used alone or in conjunction with the 4695 subwoofer.
Mfr: JBL
Circle No. 46 on Reader Service Card

DIGITAL DELAY UNIT

• The DN700 Digital Delay Unit is the first of a series of Klark-Teknik products primarily designed for engineered sound systems including theatres, conference centers, and multi-media installations. The DN700 is a single input device with three independently adjustable outputs. All operations are controlled by a microprocessor, and the delay time for each output can be varied from 0 to 435 ms. in 26.5 microsecond steps. The D.D.L. has a perpetual memory of all delay settings and features a lock-out system to prevent tampering with front panel controls. In-house designed AD-DA converters give a 15 kHz bandwidth at maximum delay, with a dynamic range greater than 86 dB. The DN700 is housed in a compact housing and can be supplied with transformer-balanced inputs and outputs.
Mfr: Klark-Teknik Electronics Inc.
Price: $1295.00
Circle No. 45 on Reader Service Card



Classified

Closing date is the fifteenth of the second month preceding the date of issue.
Send copies to: Classified Ad Dept.
db THE SOUND ENGINEERING MAGAZINE
1120 Old Country Road, Plainview, New York 11803

Minimum order accepted: $25.00
Rates: $1.00 a word
Boxed Ads: $40.00 per column inch
db Box Number: $8.50 for wording "Dept. XX," etc.
Plus $1.50 to cover postage

Frequency Discounts: 6 times, 15%; 12 times, 30%

ALL CLASSIFIED ADS MUST BE PREPAID.

FOR SALE

MCI JH24B, JH100 16 track, locator, mint $13,500.00. Electrodyne 16 x 8 console, $3,500.00. Ampex AG 440, 7.5, ½ tk., $1,800.00. **(513) 258-2463.**

Would you believe REAL-TIME cassette copies for $1.03 each? Exquisite sound, modest cost. Denon DR-F8 Recorders, AGFA Tape. Full-color printing and assembly available. America's most experienced Real-Time Duplicators. **Contact GRD (602) 252-0077, P.O. Box 13054, Phoenix, AZ 85002.**

FOR SALE: Scully "The Lathe," Nikon Scope, Ikegami camera and monitor, Ortofon DSS-732 Cutter, Custom amps, Nikon stylus inspection scope. All As New. $45,000.00. **Call (305) 947-5611, CRITERIA RECORDING STUDIOS, 1755 NE 149th Street, Miami, FL 33181.**

USED RECORDING equipment for sale. **Dan (415) 441-8934.**

THE RECORDING BOOK THAT IS ROCKING THE MUSIC INDUSTRY

"Practical Techniques for the Recording Engineer" by Sherman Keene is endorsed by major colleges, universities, recording schools, studios, musicians and our correspondence students around the world. Recommended by reviewers of the MIX, Re/P, Guitar Player and other top publications. The book—hard cover, 380 pages, 28 chapters (4 on computer assisted mixing), illustrations. $29.75 ($31.69 in Calif.) + $2.50 for shipping. The Correspondence Course—certificate course using two textbooks, corrected and graded homework, cassette dialog with the author via cassette, Basic, Intermediate, and Advanced Levels. The Curriculum—for schools only, Teacher's manual (lesson plans for two textbooks, charts, suggested session content), Student's Workbook, Final Exams. For information or to order, contact: **S.K.P., 1626 N. Wilcox, No. D-677, Hollywood, CA 90028.** Order by phone using Visa or Mastercharge by calling **(213) 708-2933.**

THE LIBRARY...Sound effects recorded in STEREO using Dolby throughout. Over 350 effects on ten discs. $100.00. Write: **The Library, P.O. Box 18145, Denver, CO 80218.**

REELS AND BOXES 5" and 7" large and small hubs, heavy duty white boxes. **W-M Sales, 1118 Dula Circle, Duncanville, TX 75116. (214) 296-2773.**

PRO AUDIO IN STOCK, ready to ship. Top lines, top dollar trade-ins, clearance specials. Call or write for our prices. **Professional Audio Video Corporation, 384 Grand Street, Paterson, NJ 07575. (201) 523-3333.**

BLANK AUDIO AND VIDEO CASSETTES direct from manufacturer, below wholesale, any length cassettes: 4 different qualities to choose from. AMPEX and AGFA mastertape—from ¼ inch to 2 inch. Cassette duplication also available. Brochure. **Andol Audio Products, Inc., Dept. db, 42-12 14th Ave., Brooklyn, NY 11219. Toll free 1-800-221-6578, ext. 1, NY residents (212) 435-7322.**

WANTED

WANTED: TRANSCRIPTION discs, any size, speed. Radio shows, music. **P.O. Box 724-db, Redmond, WA 98052.**

SERVICES

AMPEX FACTORY SERVICE: Complete service and parts for Ampex equipment; professional audio, helical-scan video systems and professional audio motor and head assembly rebuilding. **AMPEX SERVICE CENTER, 719 W. Algonquin Rd., Arlington Heights, IL 60005, 1-800-323-0692 or (312) 593-6000.**

MAGNETIC HEAD relapping—24 hour service. Replacement heads for professional recorders. **IEM, 350 N. Eric Drive, Palatine, IL 60067. (312) 358-4622.**

VIF INTERNATIONAL will remanufacture your Ampex or Scully (Ashland/Bodine) direct drive capstan motor for $200. Average turn around time 2-3 weeks. For details write: **P.O. Box 1555, Mountain View, CA 94042, or phone (408) 739-9740.**

you write it

Many readers do not realize that they can also be writers for **db.** We are always seeking meaningful articles of any length. The subject matter can cover almost anything of interest and value to audio professionals.

You don't have to be an experienced writer to be published. But you do need the ability to express your idea fully, with adequate detail and information. Our editors will polish the story for you. We suggest you first submit an outline so that we can work with you in the development of the article.

You also don't have to be an artist, we'll re-do all drawings. This means we do need sufficient detail in your rough drawing or schematic so that our artists will understand what you want.

It can be prestigious to be published and it can be profitable too. All articles accepted for publication are purchased. You won't retire on our scale, but it can make a nice extra sum for that special occasion.

EMPLOYMENT

WANTED: Audio Technology instructors with heavy multi-track industry experience needed. Candidate should possess professional skills in recording operational techniques, editing, video sweetening, etc. Immediate openings for part-time or full-time, with benefits. Send resume to: Harry Hirsch, Dir. of Education, Audio Arts Div., c/o Center for the Media Arts, 226 W. 26th St., New York, NY 10001.

Regional Sales Engineer

United States subsidiary of leading European audio company is seeking Northeastern Regional Sales Engineer to be based in New York City or, possibly, Boston area. Position requires experience with broadcast audio and/or music recording studio systems. Previous sales/management experience helpful; strong audio engineering background preferred but not essential. Must be willing to travel part time. Excellent salary potential, outstanding benefits, good chance for trips to Swiss factory. Send resume in confidence to: Studer Revox America, Inc.; Attn: Tom Mintner; 155 Avenue of the Americas; New York, NY 10013.

STUDER

An Equal Opportunity Employer

MOVING?

Keep **db** coming without interruption!

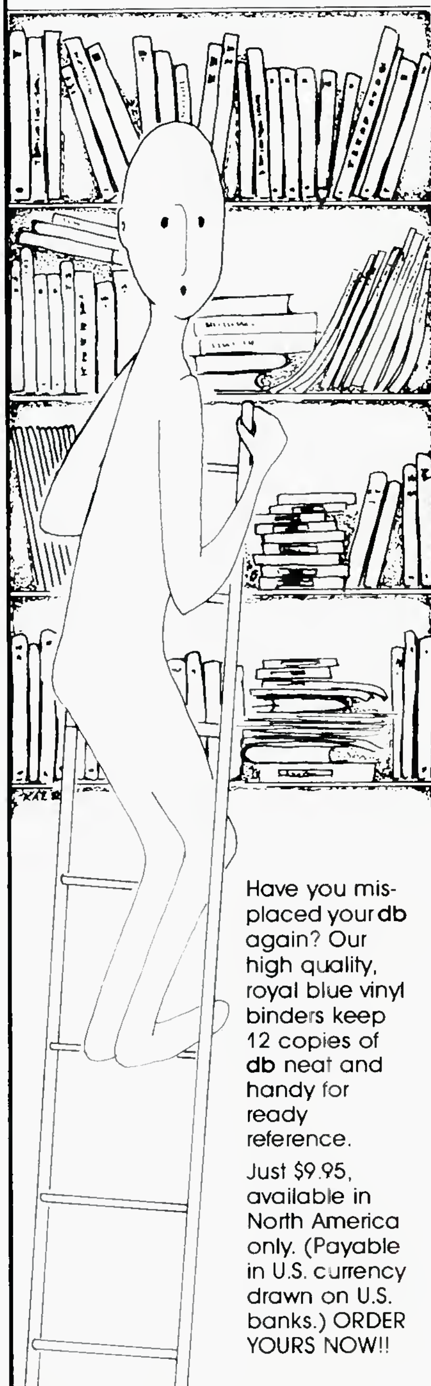
Send in your new address *promptly.*

Enclose your old **db** mailing label, too.

Write to:

Eloise Beach, *Circ. Mgr.*
db Magazine
1120 Old Country Rd.
Plainview, N.Y. 11803

UH-OH!



Have you misplaced your **db** again? Our high quality, royal blue vinyl binders keep 12 copies of **db** neat and handy for ready reference.

Just $9.95, available in North America only. (Payable in U.S. currency drawn on U.S. banks.) ORDER YOURS NOW!!

Sagamore Publishing Co., Inc.
1120 Old Country Road
Plainview, NY 11803

YES! Please send________ **db** binders @ $9.95 each, plus applicable sales tax. Total amount enclosed $________.

Name ________________

Company ________________

Address ________________

City ________________

State/Zip ________________

People, Places

• **Paul Murphy**, general manager of **Beyer Dynamic Inc.**, is pleased to announce the appointment of **Tony Hawkins** as National sales manager. Previously, Hawkins spent four years with the British Scientific Civil Service working with the Royal Air Force. He then managed the Soho branch of **Teletape of London** where he was responsible for sales of semi-professional and pro recording equipment. Later on, Hawkins worked for the **Revox Corporation** in London in various sales-related areas and was actively involved in the original distribution of Beyer product in the United States. From there, he spent six years with **Martin Audio** in New York City as a salesman/consultant specializing in microphones. During his tenure with Martin, Hawkins worked closely with recording studio architects, systems engineers and sound contractors in the design and construction of sophisticated audio-video recording facilities.

• **Brian Wachner**, president of **BGW Systems, Inc.**, has announced the promotion of **Irwin Laskey** to director of Sales and Marketing. Laskey first joined BGW Systems in 1978 as a sales representative and was later appointed sales manager. In his new capacity, Laskey will continue to develop BGW's domestic and international network of distributors and dealers specializing in BGW professional and commerical power amplifiers, as well as supervise the marketing of the newly introduced line of Professional Mobile Audio products designed for use in all mobile environments which use 12VDC power sources, such as private airplanes, boats, RVs, and luxury automobiles.

• **Steve G. Romeo** has joined the professional products division of **Bose Corporation** with the title of product specialist. He will head Bose's liaison program with acoustical consultants and sound system designers and will provide applications assistance to sound contractors and end users of Bose professional products. Previously, Romeo was chief of design for **Scenario Systems** in Denver, and president of **Destiny Light and Sound** in Boulder, CO.

• **Telex Communications, Inc.** announced the acquisition of the **Singer Audio Visual Division** of **Singer Company** of Canada, Ltd. in Scarborough, Ontario. The acquisition becomes effective in February 1983, at which time the name will be changed to **Telex Communications, Ltd.** All former Singer personnel have been retained and, in addition, Mr. **Gene Sworin** was appointed national sales manager, Canada. Telex Communications, Ltd., will continue to sell and service projection equipment and market Telex audio visual and professional audio products through dealers. This includes headsets, microphones, tape recording and high speed tape duplicating equipment for educational institutions, training in industry as well as commerical broadcasting, film and video production studios, sound reinforcement and communications.

• **Ampex Corporation** announced it has reached agreements with **Wheelabrator Financial Corporation** and **Commercial Funding Inc.** to provide financing alternatives for the lease or purchase of audio and video recorders to its U.S. customers. The new term funding program became effective January 31. The program provides customers with the opportunity to lease or purchase Ampex audio or video recorders through one of four financing alternatives: tax-oriented lease, lease purchase, conditional sale or operating lease. "The economy has necessitated the extension of financing alternatives for the purchase of capital equipment. We believe these new agreements will be a valuable asset to our customers, enabling them to lease or purchase Ampex equipment more cost effectively," said **Michael Scott**, sales finance manager, Ampex Corporation.

• **John J. Etherington**, former producer/director of **Media Concepts**, has been named general manager of **Magnetic Recorder and Reproducer Corp. Raymond F. Green**, president of Magnetic, said Etherington's appointment signals the expansion of the company into video as well as audio communications. Green is also executive vice president of **Franklin Broadcasting Company**, owner of **WFLN** AM and FM—Philadelphia's classical music stations—which purchased Magnetic in 1982.

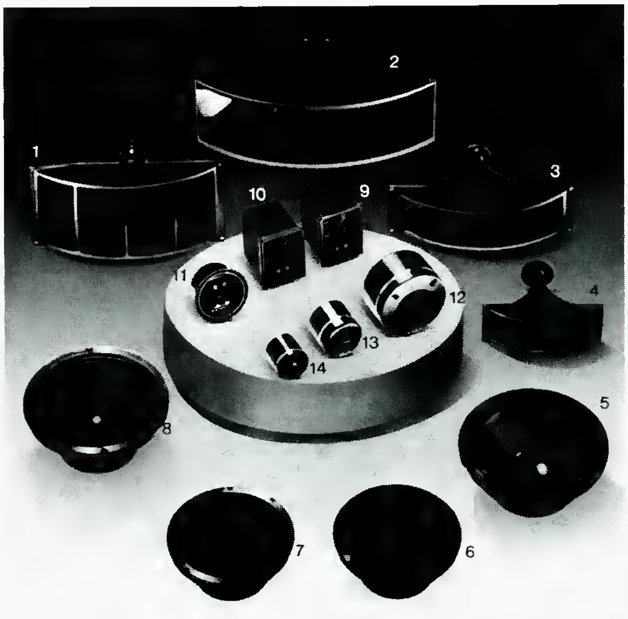
• **Soundstream, Inc.**, a wholly-owned subsidiary of **Digital Recording Corporation**, has announced a major expansion of services and personnel at its Los Angeles digital recording and editing facilities, located on the Paramount studio lot in Hollywood. Effective immediately, the Los Angeles office will become the center for the company's digital recording and editing activities. Also announced by Soundstream was the appointment of **Richard Baccigaluppi** as vice president of Marketing and Operations for the company. His responsibilities will include the broadening of the firm's digital recording/editing efforts, as well as applications of its advanced technology to better serve the needs of the audio, video, film and communications industries. **Jim Wolvington** will continue as manager of the Los Angeles editing center, and **Richard Feldman** has been promoted to recordings manager.

• The **BTX Corporation** has named **Michael L. Sipsey** vice president of Marketing, a new position in the company. Sipsey has over twelve years of technical marketing management experience and joins BTX from **Applicon, Incorporated** where he held various marketing positions including director of marketing and director of Distributor & OEM Marketing. Sipsey has also held senior marketing positions with **Nixdorf Computer Corp.** and **Entrex, Inc.**

• **John Hoge** has been appointed manager of Transducer Research and Development at **JBL Incorporated**, announced **Jerry Feingold**, vice president of Manufacturing Services at the firm. As department manager, Hoge directs the transducer engineering staff in investigating, developing, and improving component loudspeaker and system designs. In addition, he interfaces with JBL's operations and marketing divisions in formulating product development decisions and performance guidelines. Most recently an independent consultant in the areas of acoustics and noise control, Hoge has worked for such firms as **Harrison Systems, Inc., CTS Corporation** and **Studer Revox.** He is a member of such trade organizations as AES, HEEE, SMPTE, and NARAS.

Toa speaker components

give more power to more professionals.



1. **HSH-651** Sectoral Horn 2. **HRH-851** Radial Horn (large size) 3. **HRH-651** Radial Horn (middle size) 4. **HRH-311** Radial Horn (small size) 5. **HLS-3806** Low-frequency Speaker 6. **HLS-3006** Low-frequency Speaker 7. **HLS-3010** Low-frequency Speaker 8. **HLS-3812** Low-frequency Speaker 9. **HNW-351** Dividing Network (3-way) 10. **HNW-251** Dividing Network (2-way) 11. **HPF-151-8** High Pass Filter 12. **HFD-651** High Frequency Compression Driver 13. **HFD-352** High Frequency Compression Driver 14. **HT-371** Horn Tweeter

Toa speaker components are the product of years of Toa experience with sound reinforcing equipment and with the dedicated professionals who use them. To ensure that these experts have all the sound quality, efficiency and durability they've come to expect in speaker components, we've designed ours to do more. Toa offers 15 & 12 inch woofers with an edge-wound copper ribbon voice coil and high flux density ferrite magnet; radial and sectral design horns with 90° horizontal and 40° vertical dispersion; highly efficient compression driver units with smooth reproduction characteristics (power handling of 20W and 60W continuous pink noise); a tweeter with a newly developed exponential oval horn to ensure wide dispersion. And to make sure all these units work together at their best, Toa also provides an ultra-smoothly operating high-pass filter and crossover network.

The professionals at Toa put more into their speaker components so the pros who use them have the best possible audio.

Toa is also looking for professional-level dealership enquiries and OEM arrangements to help meet growing demand for the full line of high quality Toa sound products. Contact Toa for more information.

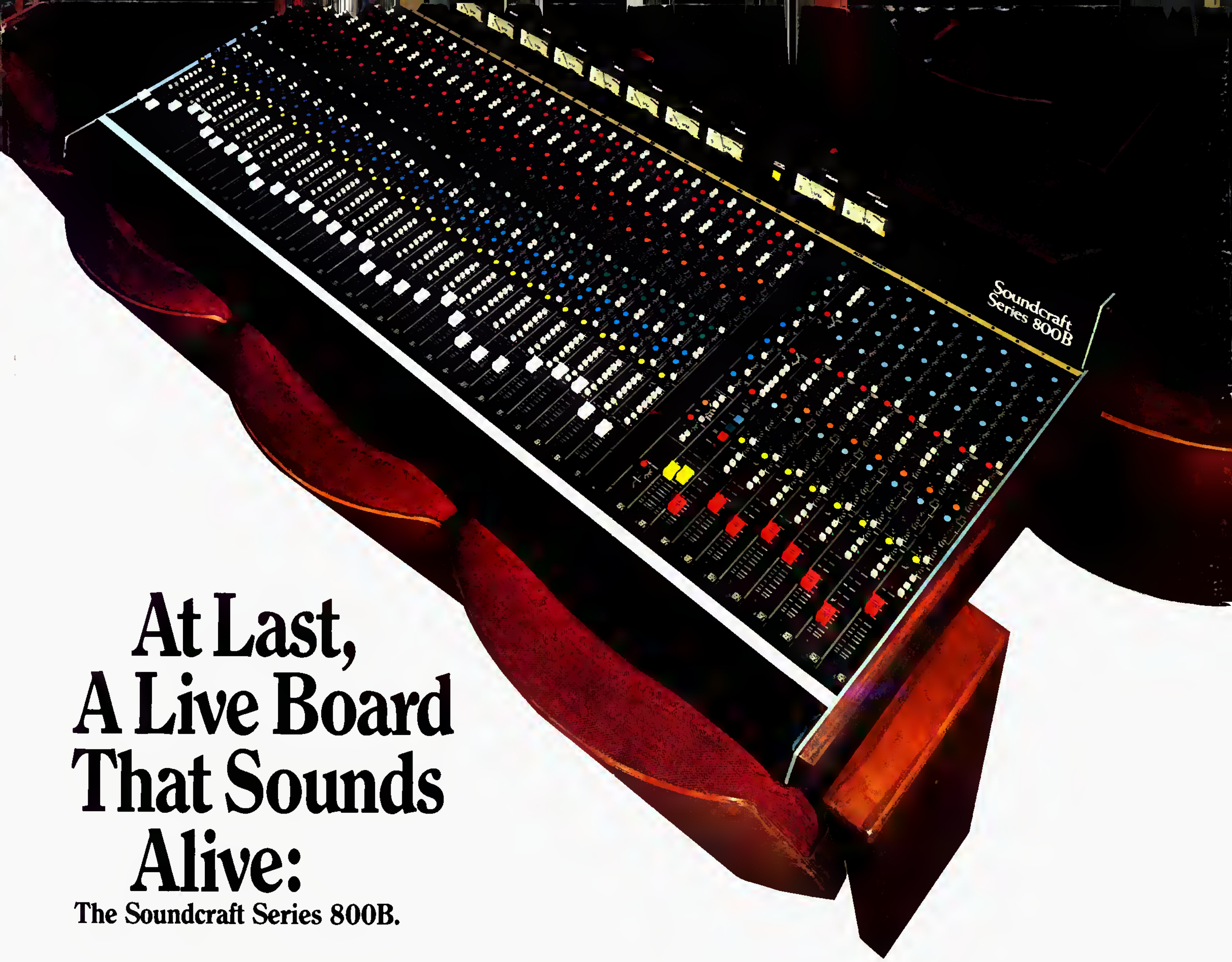
TOA Sound & Communications

TOA ELECTRIC CO., LTD.

ADDRESS:2-1 TAKAMATSU-CHO, TAKARAZUKA, HYOGO 665, JAPAN
MAILING ADDRESS: P.O. BOX 702, KOBE PORT, JAPAN PHONE: 0797-71-2211
CABLE: TOAELECTRIC TAKARAZUKA TELEX: J64150 TOA PA

• **For further information, contact your nearest Toa office:**
WEST GERMANY: Toa Electric Deutschland G.m.b.H., Kaiser Wilhelm Str. 89, 2000 Hamburg 36, West Germany PHONE: 040-345742/344213
U.S.A.: Toa Electronics, Inc., 480 Carton Court, South San Francisco, Ca. 94080, U.S.A. PHONE: 415-588-2538

Circle No. 10 on Reader Service Card



At Last, A Live Board That Sounds Alive:

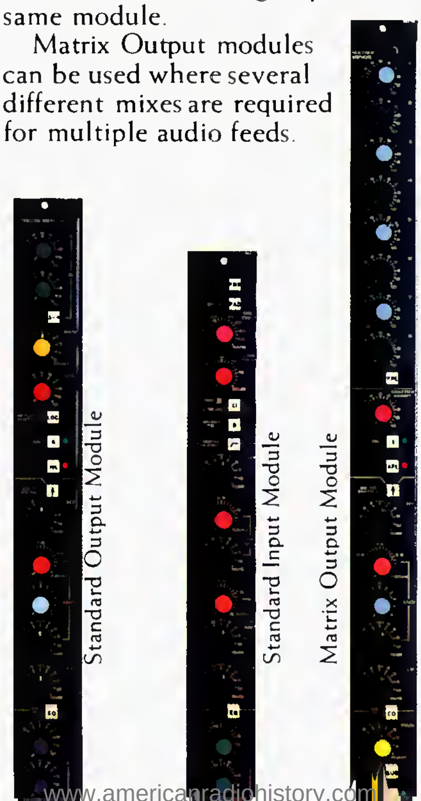
The Soundcraft Series 800B.

You expect flexibility in a live mixing board and you should *demand* sound that's *alive*. Unfortunately, that's where some of the more popular live boards fall apart. And, that's why we designed the Series 800B with headroom to spare . . . especially at the critical high end. Combine this headroom with Soundcraft's traditionally quiet electronics and you have *live sound that stays alive* through the board.

You will also find the 800B is one of the most versatile boards available, especially for its compact size. Take your choice of 16, 24 or 32 channel input mainframes with Standard or Monitor inputs . . . both with Soundcraft's famous 4-band EQ. Standard inputs feature eight Aux send busses, assignment to the eight sub-groups and the stereo mix, and long travel faders.

The Standard Output modules feature an Effects Return input with semi-parametric EQ which may be assigned to the Stereo Master or into the sub-group on the same module.

Matrix Output modules can be used where several different mixes are required for multiple audio feeds.



Standard Output Module

Standard Input Module

Matrix Output Module

To find out more about the Soundcraft Series 800B and the name of your nearest dealer, send the coupon below to Soundcraft, today. Your audience will be glad you did.

Soundcraft

Soundcraft U.S.A.
20610 Manhattan Pl. —Suite 120
Torrance, CA 90501
(213) 328-2595 TLX 182-499

Soundcraft Electronics
5-8 Great Sutton St.
London EC1VOBX England
(01) 251-3631 TLX 21198

Soundcraft U.S.A.
20610 Manhattan Pl. —Suite 120 Torrance, CA 90501
(213) 328-2595 TLX 182-499

Please send me more about the live boards that keep the sound alive.

Name ____________________

Address ____________________

City ____________________

State ____________ Zip ________