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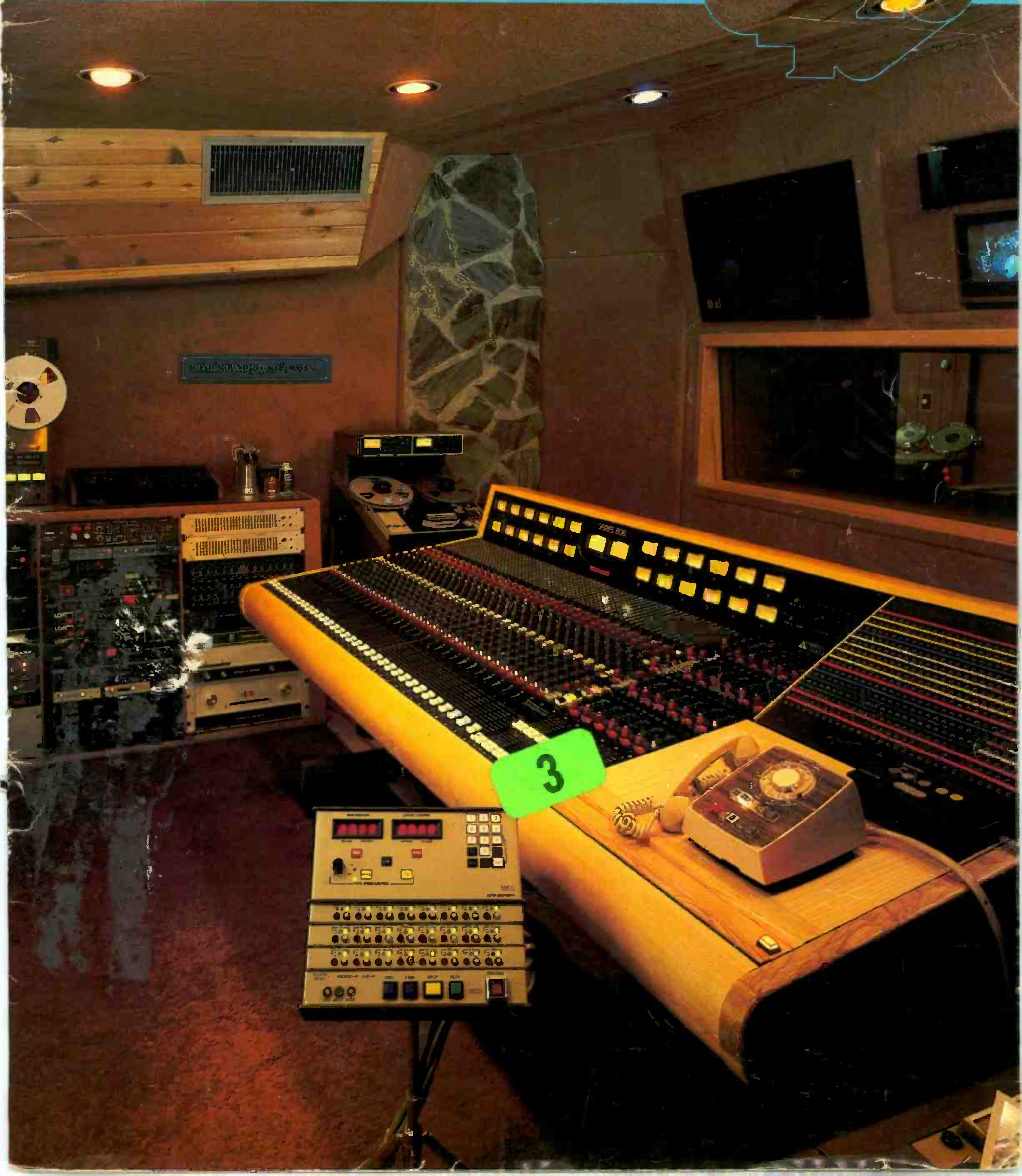
db

THE SOUND ENGINEERING MAGAZINE



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AUGUST/SEPTEMBER 1984 VOLUME 18 NO. 7

FEATURES

COLUMNS

DEPARTMENTS

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A scene from Giorgio Moroder's rescored version of Fritz Lang's 1926 film classic, "Metropolis," which includes the world's first totally digital sound track.

Letters

HEART OF GLASS

To The Editor:

I read the two excellent articles on "Glass in the Studio" by author F. Alton Everest, but as a "proprietary window" (and door) manufacturer we detected one flaw in the discussion in the May issue wherein the question of "weak windows in a strong wall" is addressed.

Specifically, the formula cited cannot be used so simply by combining Sound Transmission Class (STC) ratings of two dissimilar products.

Instead, the Transmission Loss (TL) at each of the 16 1/3 octave ratings must be plotted individually for each material to accurately determine the performance of the composite barrier.

And while the weak window/strong wall premise is appealing, it's been our experience that it makes better sense to select window and door products with equal or higher STC ratings than the wall in which they are installed.

This helps compensate for degradation that normally occurs during actual construction by failure to seal off flanking path noise, installation of electrical and mechanical equipment in the barrier and other practices.

Brian L. Williams
Director of Marketing
Overly Manufacturing Company

The point Brian Williams makes in the third paragraph is that it is better to use transmission loss at each 1/3 octave point than the general STC rating in evaluating a "weak window in a strong wall." He is absolutely right.

The concept of specifying barrier performance by a single number STC rating is a feeble simplification with many faults. But it is a simplification which the practical person appreciates, needs, and uses. The many transmission loss graphs shown in my db articles were gleaned from Canadian and German measurements. Such detailed transmission loss measurements on the specific barriers under consideration are seldom available to the average practitioner, or even to most con-

Index of Advertisers

About The Cover

• This month we feature Sonic Sound Recording Studios of Freeport, New York. Recently designed by Francis Milano of Analogique Professional Systems of New York City, the studio features a Trident Series 80 Console, MCI 24-Track Recorder, Tascam 8-Track, an Ampex ATR 102, and JBL Series 4430 Monitors. In addition, Sonic Sound uses dbx Compressor/Limiters, Lexicon Prime Time, White Equalizers, Crown Amps, and LinnDrum Computers.

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With a conservative power rating of 200 watts RMS, this new Gauss coaxial has been tested to 750 watts delivering clean sound... and can "coast" along at control room levels still delivering great sound. Metric sensitivity is 95dB for the low frequency and 109dB HF.

Because of our proprietary design parameters, both drivers are virtually in the same acoustic plane, eliminating the need for costly time compensation networks. For bi-amp operation, you can use any standard professional quality crossover.

The unique *cosh* horn was designed using Gauss's exclusive Computer Aided Time Spectrometry (CATS™) program. This design provides an extremely stable image... reduced second harmonic distortion... and virtually no midrange shadowing.

For additional information on the new Gauss coaxial loudspeaker, call or write Cetec Gauss, 9130 Glenoaks Blvd., Sun Valley, CA 91352, (213) 875-1900. Or better yet, hear it at a selected sound specialist soon.

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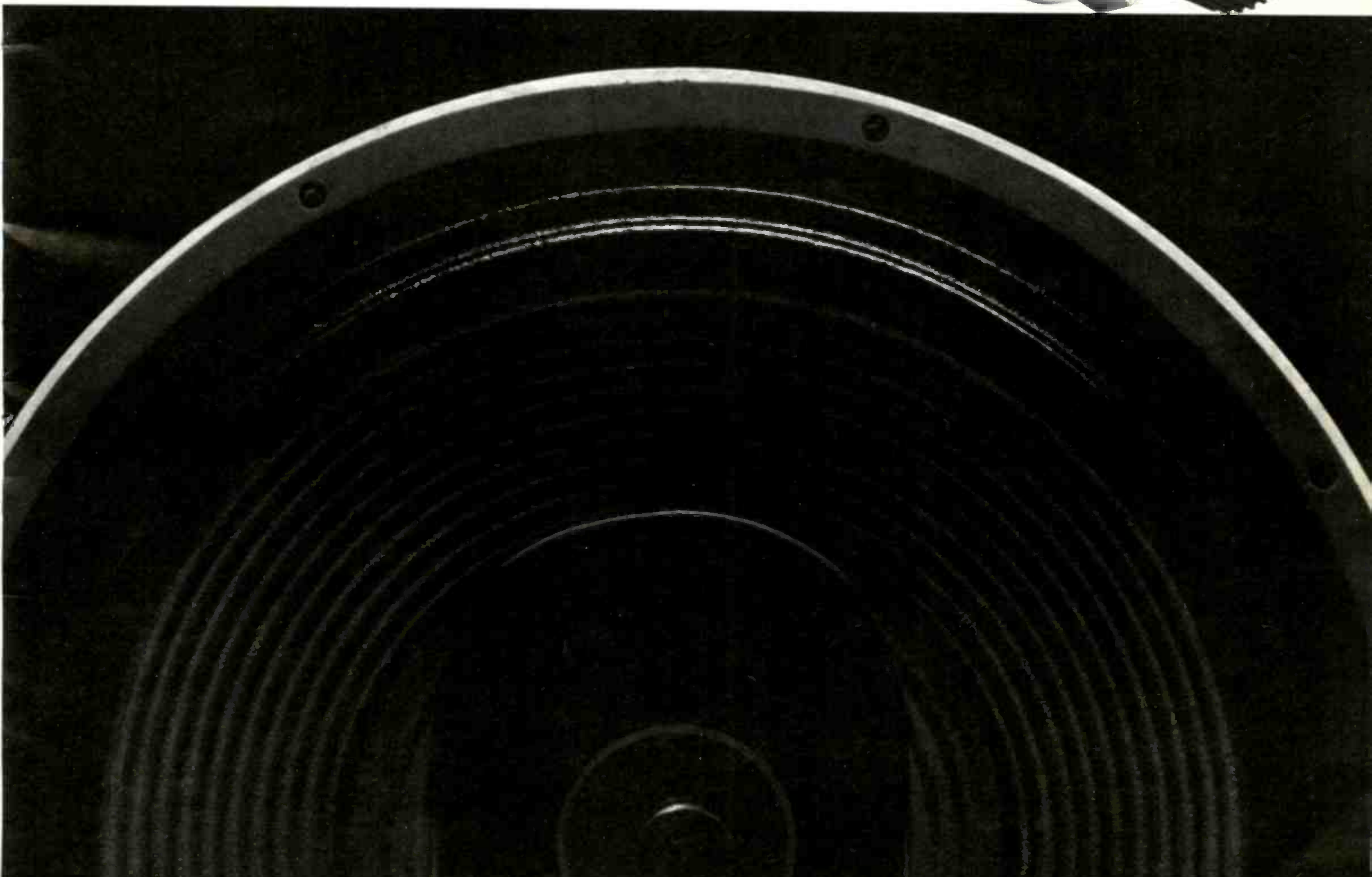

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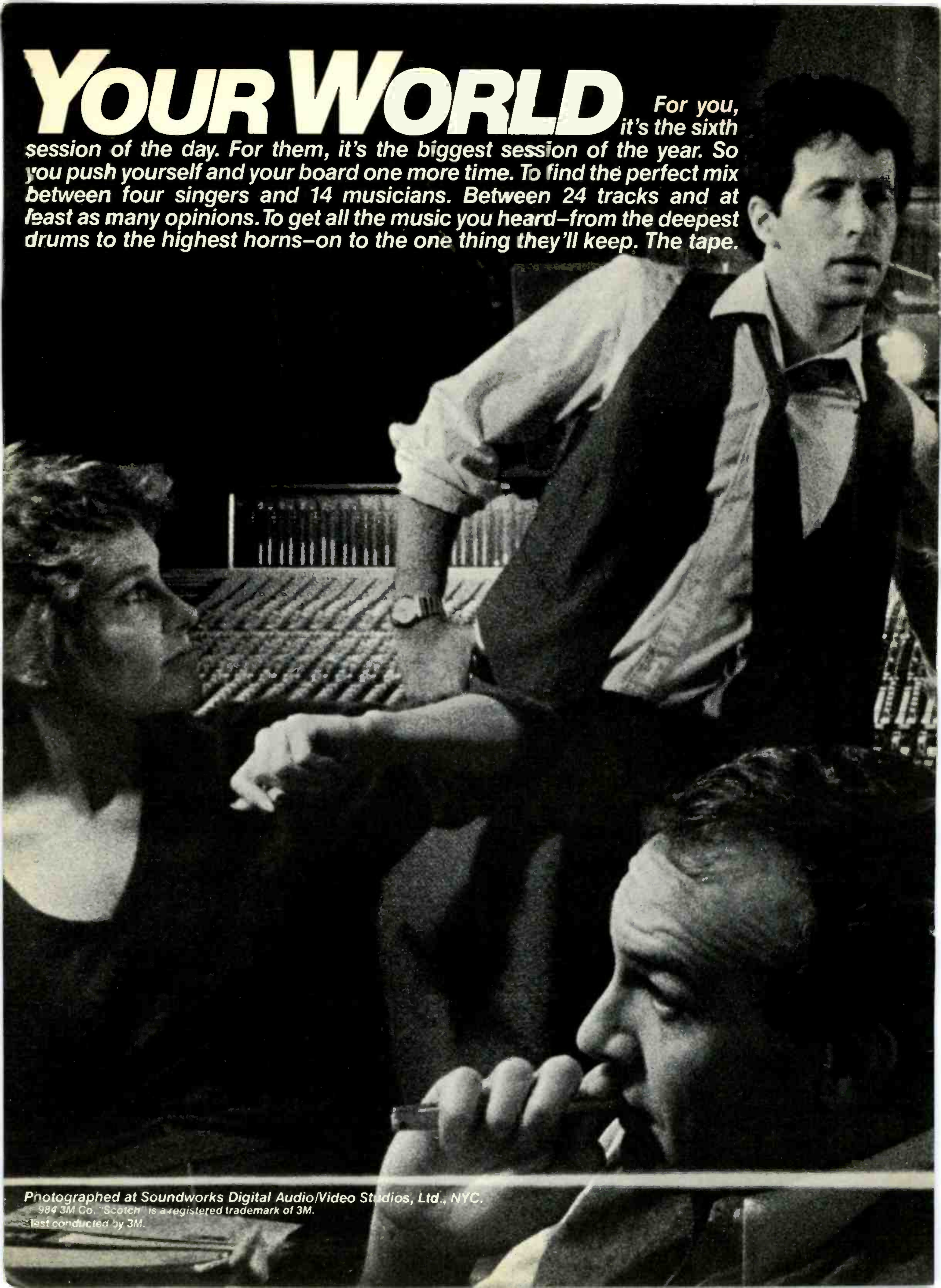
YOUR WORLD
For you, it's the sixth session of the day. For them, it's the biggest session of the year. So you push yourself and your board one more time. To find the perfect mix between four singers and 14 musicians. Between 24 tracks and at least as many opinions. To get all the music you heard–from the deepest drums to the highest horns–on to the one thing they'll keep. The tape.
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sultants. For this reason STC ratings are used which, at least, give a rough indication of what is happening. I maintain that for the purposes of illustrating the principle of a weak window in a strong wall, using STC values in EQUATION 6 *is justified. Obviously, applying* EQUATION 6 *to the transmission loss in each ⅓ octave band is far superior.*

Home-brewed windows are usually weaker than practical home-brewed wall constructions in regard to transmission loss. If transmission loss of proprietary windows available equal or exceed the desired loss of the wall between studio and control room, and the form and cost are acceptable, I fully agree with paragraph 4 of Mr. Williams' letter.

Thanks to Brian Williams for his comments.

F. Alton Everest

WANTED: ADDRESSES!

TO THE EDITOR:

I am a 1984 Electronic Engineering graduate and am seeking information on audio related companies in New England. My thesis was in audio engineering and I would like to continue in this field.

Is it possible for you to steer me in the direction of a complete list of companies, addresses, and product lines similar to those that are advertised and published in the *Journal of Audio Engineering Society*?

Whatever help you could give me would be greatly appreciated.

JOHN W. MILLER

Unfortunately, we know of no comprehensive listing of audio related companies that breaks down into regions. However, we suggest picking up a copy of Studio Sound's Pro Audio Yearbook. *For information on how to purchase the latest edition, contact Special Publications Group, Link House, Dingwall Avenue, Croydon CR9 2TA, U.K.*

The book contains an extensive listing of audio related companies (and their addresses) as well as the current products they offer.

Any readers who might know of a specific regional listing of audio companies are invited to send them our way.

WANTS TO BE C-DUCED!

TO THE EDITOR:

I found the article in the May **db** on the "C-Ducer" transducer to be very interesting. I would like to know where I can write for literature and prices of this item. I also enjoyed Bruce Bartlett's article on the TEF analyzer. Thank you for publishing these interesting articles.

ANDREW ROGULICH

Glad you liked them! For information on the C-Ducer, write C-Tape Developments Ltd., Transducer Laboratories, 73 High Street, Aldershot, Hants, GU11 1BY, U.K.

A CASE OF MISTAKEN IDENTITY

An inadvertent error in the April db caused two names in Jesse Klapholz's article "Testing...One, Two, Three" to emerge as one. There never was a Lord Kelvin Rayleigh who measured particle velocity in 1882. There was, however, a Lord John William Strutt Rayleigh and a Lord William Thomson Kelvin who did so. Mr. Klapholz's research on the two men reveals the following information:

Lord Rayleigh (a.k.a. John William Strutt) was born in Cambridge, England, in 1842. He made extensive mathematical researches in acoustics as a part of the theory of vibration in general. In 1877-78, he published, in two volumes, a treatise on the theory of sound, appropriately titled *The Theory Of Sound*, now available from Dover.

William Thomson, born 1824 in Belfast, Ireland, of Scottish descent, graduated as a Second Wrangler from Cambridge University in 1845. At 22, he was elected Professor of Natural Philosophy at the University of Glasgow. He was knighted in 1892 and made Lord Kelvin. He discovered what is called "Dirichlet's Principle" in 1848, somewhat earlier than Dirichlet did. In 1855 he predicted by mathematical analysis that the discharge of a Leyden Jar through a linear conductor would in certain cases consist of a series of decaying oscillations. He also worked on electrostatic induction in submarine cables.

The above information is from *A History Of Mathematics*, F. Cajori, McMillan & Co., 1894. P. 386. *—editor*

KEN POHLMANN

Studer Digital: A New Editing Concept

• Clearly the second wave of digital audio technology is beginning to impact our industry; not only have traditional applications begun to employ digital technology, but wholly new concepts and practices are being devised as an outcome of new avenues opened by digital processing. Audio products are more flexible thanks to the programmability and customization inherent in digital designs, and more sophisticated techniques are available at a moderate cost. In short, audio signal storage will soon be as uncomplicated (to abuse the relative nature of the idea) as computer data storage, and audio signal processing will similarly be as open-ended as computer data processing.

Audio is a technology-hungry industry; moreover, it is an industry whose practices are largely defined by its technology. Ironically, as the digital audio revolution takes root, apparently only a few companies have perceived the extent of the upheaval. The current state of thinking is fully as naive as that present during the beginning of the video recording revolution. Shortly after the development of the first video tape recorder, a committee report to the Ampex Board of Directors in 1956 estimated worldwide sales of VTRs at no more than 26 units through 1960. They did not envision the need for more tape recorders because they thought that their use would be limited primarily by television networks to compensate for time zone differences—a very limited market. Of course, that report was a little understated; in fact, by 1958 Ampex was shipping 30 units a month, and that was just the beginning of the boom. They had failed to perceive the entire idea of post production, the most important function of the video tape recorder.

RE-INVENTING THE TAPE RECORDER

I think, similarly, that digital audio technology will define entirely new market opportunities, best identified only through hindsight. Of course, some companies have more foresight than others. One

16-TRACK REALITY

Don't just dream about owning your own 16-track. Now you can get all that extra flexibility for a whole lot less than you might think.

With models starting as low as $5900*, each Fostex B-16 ½" 16-track recorder is complete with:

- Dolby + C noise reduction circuits (defeatable) on individual record/reproduce cards
- rock stable 3-motor transport mounted on a ¼" machined alloy plate
- 15 ips with ± 15% variable speed operation
- multi-pin connector for video interlock synchronizers
- Killer Sound

Why even consider a re-built old 2" machine? At two, three, even four times the price, it won't sound as good as the B-16. And it won't even perform as well as the B-16, configured with some of the options.

For example, the model with independent tape monitoring is really a whole package:

- direct drive capstan motor with phase locked loop speed control
- 7" rack mount unit with 16 independent channels of decode & reproduce (defeat switch)
- remote control unit with individual track select buttons, headphone jack and level control, line out jack and a VU meter for fast alignment

You'd have to pay almost ten times the price of a B-16 to get this kind of dedicated monitoring function. Tape reproduce is entirely separate from the record/sync electronics.

Which makes the compact B-16 perfect for live audio and video remotes. It even has handles.

And it's as easy to use as it is to own. You can expect nice user-friendly touches like:

- blinking track numbers for record ready status
- real time tape counter with search-to-zero from either direction
- servo control over reel rocking in edit mode
- spot erase capability
- coarse and fine pitch controls with blinking LED for ON status
- optional full function remote control and auto locator

Increase your audio production capability while decreasing your costs. You'll not only save on your initial investment, but operating costs as well — both tape and maintenance.

Right now, the B-16 is the smart move in 16-track hardware. Let your Fostex Professional Multitrack Dealer° prove it. For real.

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company, in my opinion, stands in the forefront of the second wave of digital audio technology, particularly in terms of perceiving its eventual impact, and meanwhile guiding its course. Studer has recently presented several papers outlining some of its vision of digital audio technology. Studer has examined digital technology on its own terms, freed of analog preconceptions. For example, instead of designing a digital tape recorder as an analog machine with digital circuitry, Studer has attempted to re-invent the tape recorder using concepts specifically promoted by digital technology. The result, as we saw last time, is a recorder which is an extension, rather than a reworking, of past analog designs.

A recent paper authored by Dr. Roger Lagadec of Studer pointed to another direction for digital audio technology. Editing is a primary concern of audio post production, and just as the advent of the video tape recorder and editing systems opened up entirely new practices, digital audio editing systems could similarly revolutionize the audio industry. As Dr. Lagadec points out, traditional cut-and-splice methods worked well for analog recording. However, those methods will have to be modified and supplemented by new methods for efficient editing of digital audio tape. Tape cut editing can be accomplished very quickly, but data is destroyed and it is difficult to incrementally re-cut an edit point with digital tape. Electronic editing preserves data and offers unlimited rehearsal. However, either multiple tape machines or a disk pack are required, both contributing to cost overhead. Given a digital tape recorder with read-after-write capability, and self-sufficient ability to identify individual samples, several editing alternatives are possible, including at least one hybrid method which uses data labels to combine the advantages of tape cut and electronic editing. In other words, a combination of sequential and random access techniques might provide the most effective immediate solution.

A digital tape recorder interfaced to external storage offers one technique for digital audio editing; systems using disk packs have proved successful. However, the majority of studio applications do not require an hour of on-line random access audio data. Thus the use of smaller disk systems becomes a possibility. Hard disk systems presently lack data transfer rate and access time for satisfactory operation in audio applications. Future magnetic disks will undoubtedly feature better performance specifications, and Winchester type drives would be ideal for audio editing. Also, optical disk DRAW systems may soon be incorporated into an editing system; they promise to greatly increase storage density. In addition, decreases in the cost of solid-state memory and advances in floppy disk storage might permit those formats to be used to store small amounts of audio information, sufficient to accomplish editing.

PROBLEMS, PROBLEMS

Since those techniques must wait for technological advances before they may be effectively introduced into the studio environment, and since presently available electronic editing may not be immediately cost effective, our attention naturally returns to tape cut editing. With digital audio, several difficulties present themselves: Data is destroyed around the splice point and its interleaved continuity lost; thus, interpolation must be employed to

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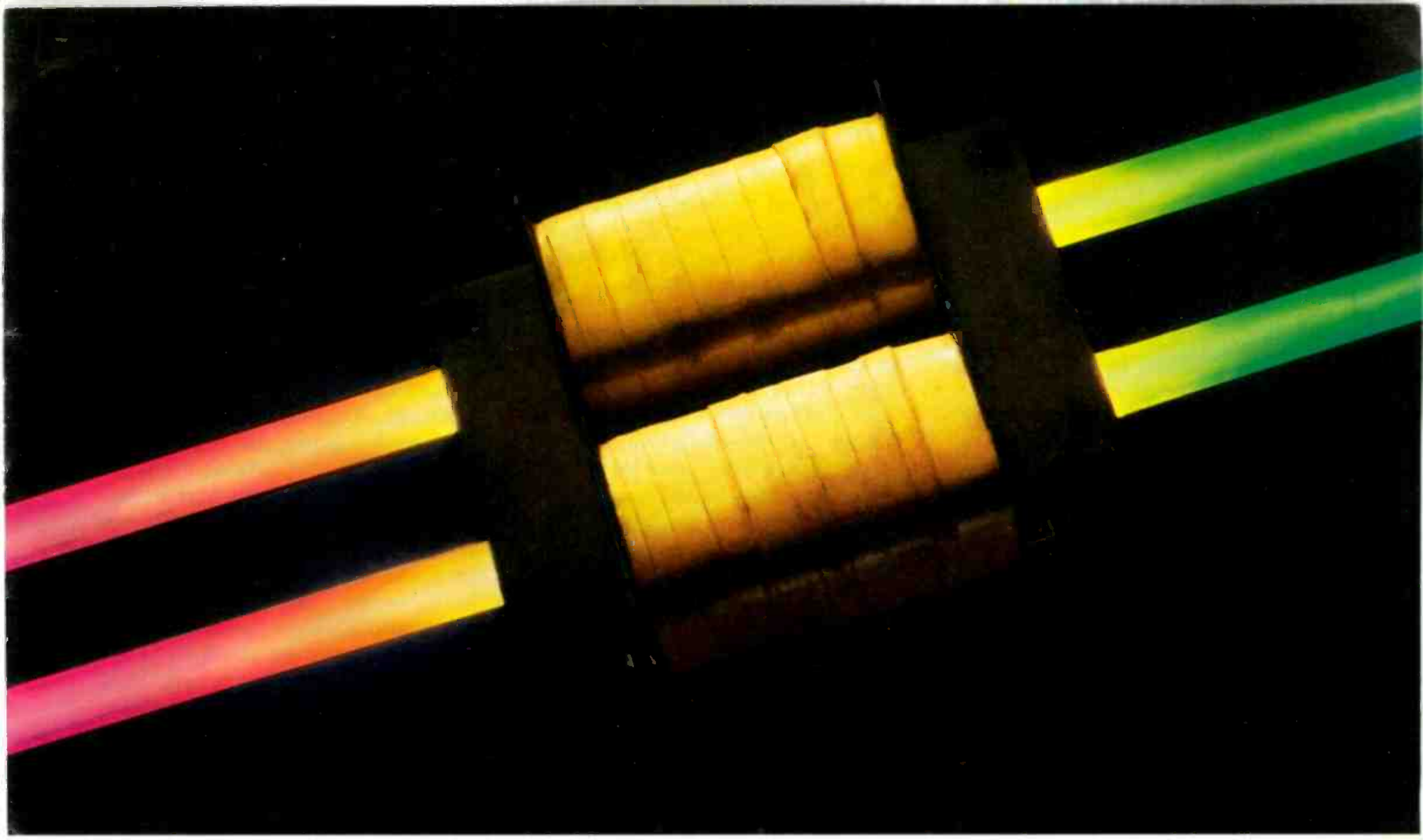

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A vacuum tank, to be precise. An environment in which our transformer is created with an air expulsion process. Where air pockets are purged from multi-layers of the transformer's high grade core laminations, and the core made airtight with a special impregnating sealant.

This process gives us impeccable control of transformer function and quality. Which gives you the assurance of the most efficient transfer of power possible, and an end to the acoustical "buzz" that so often fights sound purity.

To a lot of manufacturers the lowly transformer is far down on the list of priorities. For us, every element in the sound system relies on the exacting performance of every other element, and must be painstakingly attended to.

From ultimate accuracy in laying down your initial tracks, to capturing the full power and subtlety of your final mix, you'll find our technology giving outstanding clarity to your work in products from our 6000 Series amplifiers to our limiter/compressors and a broad line of other signal processing equipment. To find out which system meets your needs, contact your authorized JBL/UREI professional products dealer today.

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reconstruct the bit stream. Incremental cutting is difficult, and errors often accumulate because of the thinness of digital tape. For example, Ampex 456 is 2.02 mils thick and 466 is 1.24 mils thick; 3M 250 is 2.06 mils and 265 is 1.05 mils thick. Another disadvantage is the simple cost of cutting expensive digital tape and rendering it useless for other applications. Finally, cross-fading requires external processing with digital tape cut editing.

However, Lagadec offers an elegant solution that rescues digital tape cut editing from some of these disadvantages while preserving its essential quickness and simplicity. While the definition of intelligence might be argued, it is safe to say that a digital recorder is smarter than an analog one, thus it is possible to abandon the dumb technique in which a cut point always coincides with an edit point. If edit points were removed from the damaged cut point, the recorder could jump over the cut point, preserving the bit stream. To accomplish this, the recorder must know when to jump, how far, and how to cross-fade; a buffer long enough to accommodate the edits must be provided. For example, with the DASH-S format, one edit would require about 0.4 seconds of data storage.

LABELS

The secret of supplying jump information to the recorder comes in the form of editing instruction labels written on an auxiliary track. (As we will see in the future, Studer holds great hopes for formatted user data to enhance machine smartness.) Thus, jump editing is controlled from information held in labels written on the auxiliary data track; since they are time-coded to the audio data, a jump label would merely have to be recorded in the vicinity of the cut point. The edit labels would identify the edit points in the audio data on either side of the splice, and control cross-fading and gain-changing. Editing labels would be one of many types of labels on the auxiliary data track; other applications would include operational data such as program duration, take number, and cue information, technical data such as audio wordlength, pre-emphasis, compression, and equalization.

Of course, writing labels on the auxiliary data track is only half the battle; buffer size and its introduced delay must be considered. Prior to editing, an edit buffer would operate as a pure delay line in the playback chain. For example, a one-edit buffer would introduce about 0.4 seconds of delay. After accomplishing the jump, the buffer's capacity would be exhausted and it could not accommodate another jump; memory underflow would occur. One solution is a very large buffer able to accommodate many edits (and introducing a long delay). A more cunning approach is to successfully re-fill the memory by increasing the speed of the tape such that data would be fed into the buffer at a rate higher than that at the clocked output. After an edit, the transport would momentarily leap ahead, and at some point the buffer would again be full and ready to accomplish another jump. Labels would thus also be counted on the manipulate transport speed. Data equalization would have to be considered, and buffer length would be dependent on number, spacing, and duration of jumps, and the dynamics of the transport. Additionally, a warning sign would have to be prominently placed on the tape machine so that uninitiated users would not freak out.

JUMP EDITING

A system to accomplish jump editing would not be overly complicated, and the ease of use may be illustrated by a possible operational procedure for performing a jump edit:

1. Locate the approximate position of the upstream edit point.
2. Tape is automatically positioned to be cut after the edit point.
3. Locate the approximate position of the downstream edit point.
4. Tape is automatically positioned to be cut before the edit point.
5. Perform tape cutting and splicing.
6. Labels are written on the auxiliary data track.
7. The edit is rehearsed.
8. If required, the edit point is moved by modifying the labels.

Depending on the size of the internal buffer and the sophistication of the system, the editing could be largely automated and rehearsed without moving the tape. Alternatively, the ultimate simplicity and low cost of razor blade editing could be maintained with manual operation. The point is that a tape edit can be quickly performed, then rehearsed, time shifted, and finalized by simply re-writing jump labels. Thus, splicing is primarily automatically performed, with only one rough cut. If there was need to perform editing within one error code interleave length, a supplemental random access memory could be used.

Jump editing is thus potentially an alternative to tape cut, disk-based, and tape electronic editing techniques. It is fast, data degradation-free, and requires only a single smart recorder. More importantly, I think, jump editing is an example of smart thinking on the part of companies unafraid to capitalize on new technology to re-define practices. That's a lesson we should never forget: the true utility of technology lies not in its own sophistication, but in its imaginative application. ■

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Hammar, P. "Origins of Video Tape Recording." *Mix Magazine*, August 1984.

Lagadec, R. "New Concepts in Digital Audio Editing." 75th AES Convention, Paris, March 1984.

Lagadec, R. McNally, G. "Labels and Their Formatting in Digital Audio Recording and Transmission." 74th AES Convention, New York, October 1984.

Headware's Acoustics II Software Package

• Put a half dozen sound engineers in a room, give them the parameters for a sound system that you'd like to have specified, and you'll get at least two dozen different designs. How is one supposed to create a software package to accommodate every whim and fancy of every sound engineer? Everybody has a different approach when it comes to sound system design, but the one thing we all have in common is the laws of physics, by which we are all equally bound.

Using a computer as a tool in the sound system design process can greatly reduce design/computation time, fully document design computations, reduce mathematical errors, and graphically present the data in various "what if" conditions. However, a computer will not design a sound system for you. Anyone who believes that a computer will "think" for them is unfortunately misguided.

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SECTION 1 - AXIAL MODE STUDY
SECTION 2 - REVERB TIME ANALYSIS
SECTION 3 - HELMHOLTZ RESONATOR DESIGN
SECTION 4 - TEMPERATURE, SPEED OF SOUND, FREQUENCY, & WAVELENGTH
SECTION 5 - BOUNCE PATH ANALYSIS
SECTION 6 - RECALL GRAPHS/PLOTS
SECTION 7 - CREATE ABSORPTION COEFFICIENT FILES
SECTION 8 - VERIFY ABSORPTION COEFFICIENT FILES
SECTION 9 - DELETE ABSORPTION COEFFICIENT FILES

Figure 1. The modules that make up the Acoustics II program.
```

Figure 1. The modules that make up the Acoustics II program.

Computer programs that are designed without restriction to a set flow-chart design process make most effective tools, allowing the program to be adapted to the designer, rather then have the designer adapt to the computer program. In this month's column we'll be taking a close look at one of these modular type programs.

Acoustics II is a comprehensive

```
H : W : L        DESCRIPTION
------------------------------------
          1.26
1   1.5   2.5     SABINE

1   1.6   2.5     VOLKMANN

1   1.62  2.62    GOLDEN SECTION

1   1.67  2.67    EUROPEAN

1   1.97
1   2     3       HARMONIC

PRESS SPACE BAR TO CONTINUE
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Figure 2. Ratios of a room in comparison to accepted ratios.

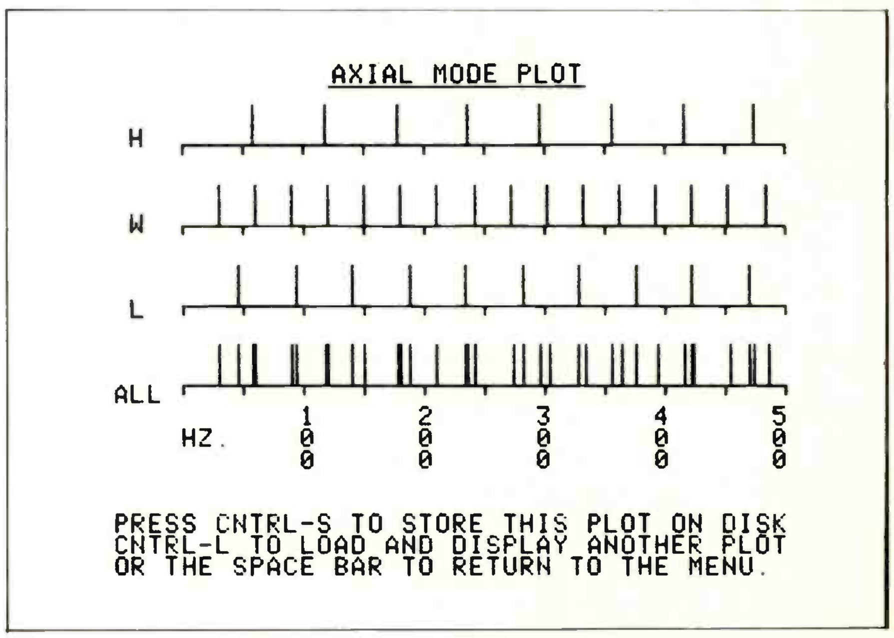


Figure 3. Axial mode plot graphic display of calculated Eigen modes.

software package that is available from Headware[1], written to run on the Apple II computer. (An IBM PC version will be available soon.) The program consists of nine interactive modules as listed in FIGURE 1. Upon loading the program into the computer, you may execute any of the nine modules. The program has a default value for the velocity of sound —i.e., a preset value if no other value is selected—of 1130 feet-per-second. However, if the Temperature, Speed of Sound, Frequency, & Wavelength is selected first, the values entered will be used for any subsequent computations with the other modules, allowing for exact calculations of wavelengths and frequencies.

Perhaps one of the most useful features of Acoustics II is that any screen display may be printed on a dot-matrix printer with a single keystroke command. This "screen-dump" feature produces complete graphic/text documentation of the entire program design process (and/or any absorption coefficient file), which is more than adequate for both "in-house" use and client presentation.

REVERBERATION TIMES IN SECONDS :

FREQ.	SABINE	EYRING	FITZ(S)	FITZ(E)
128	.68	.59*	.71	.62
256	.41	.32*	.67	.58
512	.4	.31*	.69	.6
1024	.29	.2*	.43	.33
2048	.25	.15*	.32	.21
4096	.25	.15*	.3	.2

* = PREFERRED CALCULATION.

PRESS 'G' TO GRAPH PREFERRED RESULTS,
CNTRL-E TO REVIEW DATA, OR THE SPACE
BAR TO RETURN TO THE MAIN MENU

Figure 5. RT60 display with 'preferred' times marked.

The Axial Mode Study is the first module of the program and aids the designer with quick analysis of the spectral distribution of axial modes (or standing waves as they are more commonly referred to). The computer prompts (asks for information to be input) you for the room dimensions in feet and inches and displays your answer in decimal notation. (A version of the program in metric units is in the works.)

Pressing the space-bar after inputting the room dimensions will display their ratios. For easy comparison, these ratios are displayed next to various accepted ratios that yield a desirable spread of axial modes. Pressing the space-bar again displays the first 16 modes in each dimension. The computer then sorts all of the modes in ascending order of frequency, which may be displayed by pressing the RETURN key. This new display also shows the difference in frequency between successive modes, with differences of 2 Hz or less, or 20 Hz or more, being highlighted for quick easy display of trouble spots due to spectral grouping of axial modes.

At this point in the modal analysis there are several options:

1) Re-run the Axial Mode Study module.
2) Return to the main menu.
3) Enter directly into the Reverb Time Analysis, using all the room dimensions entered in the Axial Mode module.
4) Graph the results from the Axial Mode Analysis in one display.

END WALLS DATA OK? (Y/N) Y
SIDE WALLS DATA OK? (Y/N) Y
FLOOR DATA OK? (Y/N) Y
CEILING DATA OK? (Y/N) Y
HIT SPACE BAR TO CONTINUE

SURFACE:	128	256	512	1024	2048	4096
	- - - - - FREQUENCY - - - - -					
END WALLS	.19	.08	.07	.14	.22	.24
SIDE WALLS	.34	.56	.6	.68	.77	.72
FLOOR	.02	.06	.14	.37	.6	.65
CEILING	.43	.83	.77	.97	.99	.96

Figure 4. RT60 module display of room surfaces and absorption coefficients.

WHAT HAPPENS IF WE...?

Many "what if" scenarios may be played out in a matter of minutes, with complete graphic documentation for easy comparison of, "What if we move this wall...or change the ceiling here...?". Standing waves (especially in small rooms) are a fact of life; there is not much at all we can do to totally eliminate them. However, being able to spread them evenly across the low frequency spectrum will yield a smooth-sounding bottom end in the room. This is not only applicable to control room design, but sound reinforcement applications as well. Standing waves account for many low-frequency ring modes, resulting in feedback which can only be partially cured by narrow band equalization.

REVERB TIME ANALYSIS

The Reverb Time Analysis module is one of the Acoustics II modules that everybody should be able to use. The two attractive features of this section are: first, it uses all four accepted RT60 formulae (Sabine, Eyring, Fitzroy/Sabine, and Fitzroy/Eyring), automatically optimizing the curve in six octave-bands using the formula most suitable in each of the octave bands. Secondly, the module uses a library of absorption coefficients of 80 different materials (the program comes with 40, and you may add up to 140 more).

The tedious tasks normally involved with the manual computation of RT60 (either with a hand calculator or a slide rule)—calculating the areas of the various room surfaces, looking up the numerous absorption coefficients, calculating the absorption coefficients of each surface, and finally calculating the RT60—THEN REPEATING THIS ENTIRE PROCESS SIX TIMES—are all done in a matter of minutes, with graphic printouts available of every step in the process.

As in all the other modules, in the Reverb Time Analysis module, the computer prompts you through the entire data input process, and asks you how many materials are on each surface, what materials each surface is comprised of, and the surface area that each material covers. The program then tells you the percentage of surface area that each material covers, and displays a chart of the calculated absorption coefficient factors, in six octave-bands, for all the room surfaces. Once the room constants have been established, the reverb times may be calculated using all four formulae applied to the six octave-bands, and are all displayed simultaneously, with an asterik(*) "flag" next to the "preferred" answers found to be most reliable (usually Sabine or Eyring).

After all the reverb times are calculated, the "preferred" RT60s may be plotted on one of three different automatically selected time scale graphs: 0 to .5 seconds, 0 to 1 seconds, and 0 to 5 seconds. If any reverb time exceeds five seconds, it is plotted at the top of the scale as if it was a five-second reverb time. Five seconds is the maximum RT60 that can be calculated, since the formulae are optimized for RT60s of five seconds or less.

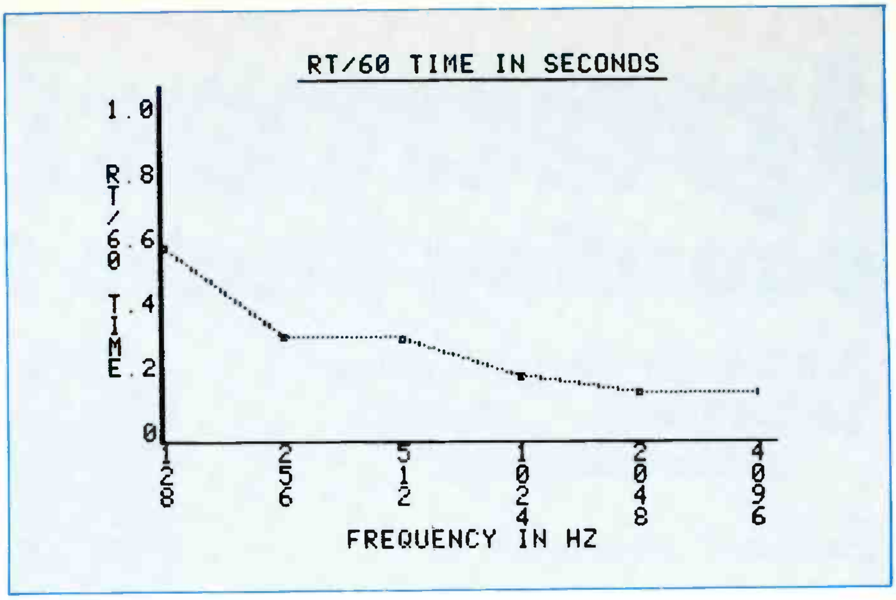


Figure 6. Graph of above 'preferred' calculations.

TEMPERATURE, SPEED OF SOUND, FREQUENCY, AND WAVELENGTH

The Temperature, Speed of Sound, Frequency, & Wavelength module, as the manual refers to it, is, "...a very simple 'acoustical calculator'." The three variables to the simple equation F = VW (where F = frequency, V = velocity of sound in air, and W = the wavelength) is what this module is all about. Although it is such a simple conversion to do manually, as mentioned earlier, the feature of having the value of the speed of sound entered in this section and then used in the rest of the program is convenient in producing more accurate results throughout the entire program.

BOUNCE PATH ANALYSIS

The Bounce Path Analysis module of the program makes a good companion to the Axial Mode Study module. This module facilitates analysis of the frequency-dependent effects of reflections within a space. Even though the program is primarily set up to deal with first-order reflections, second-order or greater "bounce paths" may be analyzed, keeping in mind that the intensity of a sound wave is reduced with each successive reflection along with the interference effects. This section can be a great tool in looking at the effects of room geometry, the absorption of room surfaces upon the directional, spectral content, and intensity characteristics of a sound source in a room design.

```
A) ACOUSTIC PANELS -3.5"-5.4 LB.
B) BRICK - WALL
C) BRICK - WALL - PAINTED
D) CARPET - HEAVY & ON CONCRETE
E) CONCRETE - COARSE BLOCKS
F) CONCRETE - PAINTED BLOCKS
G) CONCRETE - SMOOTH WALL
H) CORK FLOORING - 3/4"
I) FIBERGLAS - MAT FACED - 1"
J) FIBERGLAS - MAT FACED - 3/4"
K) FIBERGLAS - MAT FACED - 5/8"
L) FIBERGLAS - PIN PERFORATED 5/8
M) FIBERGLASS - 1" PAINTED TILE
N) GLASS - HEAVY PLATE
O) GLASS - MIRROR
P) GLASS - WINDOW
Q) GYPROCK - 1/2" ON 16" STUDS
R) LINOLEUM - FLOOR TILES
S) MARBLE
T) MASONITE - 7/16"

LETTER OF MATERIAL TO READ?      (PG. 1)
```

Figure 7. Twenty of the library of the forty materials that are included with the program.

The Bounce Path Analysis module prompts you to input the name of the sound source, the length of the direct sound path, the number of bounce paths to be analyzed, the names of each bounce point, and the bounce path lengths. Now with the simple press of a button, an individual display is created for each bounce point showing: the name of the source and bounce point, the path length difference between the direct path and bounce path, the full wavelength of this difference, and the delay time of the arrival of this reflection. Next, the display shows the predicted peaks and dips in the frequency response expected at the listening position, due to the path length differences.

After all the bounce paths have been analyzed, the program will automatically display a plot that indicates the center frequency of each peak and dip, calculated for each bounce point input. The intent

```
THIS ROUTINE WILL READ AN ABSORPTION
COEFFICIENT FILE AND DISPLAY ITS
CONTENTS
----------------------------------------

FILE - ACOUSTIC PANELS -3.5"-5.4 LB.



            128 HZ =   .6
            256 HZ =   .97
            512 HZ =   .97
           1024 HZ =   .93
           2048 HZ =   .91
           4096 HZ =   .78


HIT 'M' FOR MORE, OR THE SPACE BAR TO
RETURN TO THE MAIN MENU
```

Figure 8. A sample display of one materials' absorption coefficients.

of the program is to design for an even distribution of peaks and dips, in addition to avoiding overlaps with modal problems.

HELMHOLTZ RESONATOR DESIGN

Using Helmholtz resonators in rooms is a widely used and accepted technique covered by many texts and articles. Perforated panel absorbers and slat type absorbers are the most popularly implemented designs, and are the two types that can be designed with the aid of the Helmholtz Resonator Design module of the program.

If the perforated panel absorber is selected, and the panel thickness, hole diameter, hole spacing, and the geometrical relationships of the holes are input, and the program promptly responds with the required cavity depth to achieve resonance at the desired frequency. If slat absorbers are selected, then the slat thickness, slat width, and the available cavity depth are input. The program then calculates the width of the slats for the absorber to operate at the desired frequency. If the specified cavity depth is not sufficient for the slat absorber to resonate properly at the specified frequency, the display will read, "INCREASE CAVITY DEPTH."

USER FRIENDLY... IDIOT PROOF

No matter how hard I tried, I couldn't get the program to "crash," and errors associated with user input seemed to be thought out and avoided in its structure. Acoustics II is easy to use and does not require more than a basic working knowledge of computers. However, it does require—as any Computer Aided Design program for sound or acoustics—a good understanding of acoustics and room design.

```
      BUMP ANALYSIS - RIGHT WOOFER
      ------BOUNCE PATH - R1------

THE PATH LENGTH DIFFERENCE IS 3.25 FT.
THIS IS ONE FULL WAVELENGTH AT 348 HZ.
THE REFLECTION ARRIVES AFTER 2.9 MSEC.

---------------------------------------
 F/2    F     3F/2   2F    5F/2   3F
---------------------------------------
 DIP   PEAK   DIP   PEAK   DIP   PEAK
---------------------------------------

 174   348    522   696    870   1044

---------------------------------------

INTERFERENCE FREQUENCIES ABOVE ARE THE
CENTRES OF THE PEAKS AND DIPS, IN HERTZ

PRESS SPACE BAR FOR NEXT BOUNCE PATH
```

Figure 9. Display of first path in the bump analysis module.

Acoustics II can be classified as what is known as "user friendly." That is, it walks you through the entire procedure with plain English requests for input of information or selection of options, and waits for an acceptable response. As was mentioned earlier, any screen display can be printed out on a standard dot-matrix printer in a matter of seconds upon a single keystroke command. These printouts put the comparison of many "what if's," combinations of design elements, and complete documentation at your fingertips. Anyone who has done any of these acoustical computations manually should find this program to be a great time, brain, and eraser saver.

CAD is the acronym for Computer Aided Design. The key word here is "Aided." A computer can be an invaluable tool in the design process. If a structured flow chart design process was set up and adhered to in a computer program, designs taking advantage of practical experience and intuition could be overlooked. As we have seen, Acoustics II does not follow a structured flow chart design process; instead, it allows itself to be incorporated into your own flow chart.

There is no analyzer or computer that can be interfaced into a space and/or sound system that will analyze and produce a readout of all the errors and faults in that space and/or system. A computer is a device that can help us if used as a tool, but that can get us into trouble if taken as the "gospel."

Just as the readout on a simple hand-held sound-level meter is useless information to the typical night club owner—since he has little knowledge of the decibel notation system or the basic laws of acoustics—the readout of a computer program is useless data to the person who doesn't fully understand the acoustical principles which the program itself uses. The more complicated (and hopefully more powerful) a program is, the more necessary a knowledge of the physical laws involved become. ■

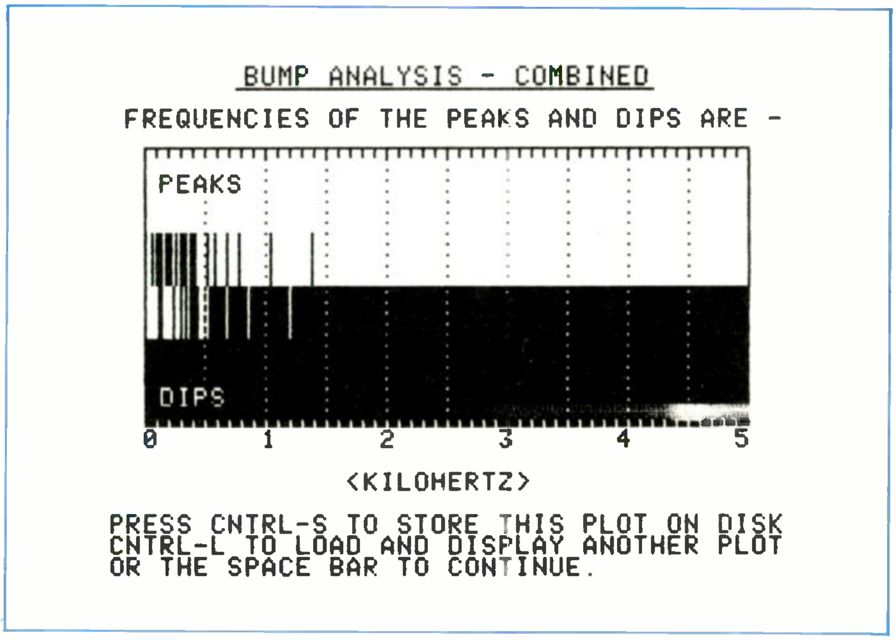


Figure 10. Graphic display of six paths from single source, showing resultant peaks and dips.

REFERENCE

1. Headware is a new Canadian software publishing company specializing in microcomputer software for the recording and sound industry. They can be contacted at P.O. Box 1106, Station F, Toronto, Ontario, Canada M4Y 2T8. Phone: 416-731-2496.

BARRY BLESSER

Why Bother?

• There comes a time in every activity when we must ask why we bother. Each month for the last four years I have sat down at my typewriter to select an interesting topic on digital audio. And, each month, many of you spend a few minutes to read what I have written. Perhaps it might be interesting to examine what I have achieved over these years. Have I been successful? Since I have not received any direct communication from the readers, I can only speculate that I have been either very successful or a total failure. I would like to take a few minutes to write about what I had hoped to achieve, and we can then evaluate my performance.

Although I have had extensive design experience in digital audio, these articles in **db** are not intended to make the reader a design engineer. Although digital audio has its roots in higher mathematics, the **db** articles are not intended to provide a formal education. The articles are, however, intended to give the reader a *feel* and *intuition* for the subject. Having good intuition about the subject is not the same as having formal training. Curiously enough, there are often people with formal training who have no feel for their subject and people without formal training who do have such a feel.

How does one get a feel for a difficult subject? Next time you are at an AES convention, go take an old-timer out for a free beer and ask him to talk about how he learned audio in the good (bad) old days. The stories always have a common idea: play with it and experience it through your senses. He will invariably talk in terms of senses: touch, sight, hearing, and smell (when something overloaded). The grooves of a record are visible under a microscope; the filaments of the tube glowed red hot; the vibrations of the loudspeaker cone could be felt by the touch of the hand; even the harmonic distortion could be clearly heard when grossly overdriving the amplifier. Even the sophisticated equipment had a physical and sensory component: you could see and touch the reverberation plate, you could see and touch the microphone ribbon, you could see and touch the contacts of the fader. Wires could all be visualized as pipes containing the flowing music. Even the magnetic fields on recorded tape could be made visual with the aid of magnetic fluid.

Another element of the old-time stories is that the audio people never worked at their profession; they were always having fun playing. Learning and doing was always fun. They would say "What happens if I...?" And then they would go do it to see what happens. When I was a kid, I learned about the differences in vacuum tubes by breaking them open and taking them apart. I had the largest collection of grids and plates on my block.

Then came the revolution!

THE REVOLUTION

Now that the revolution has started sweeping over us, we can look back and see that it actually started in the early 1960s. We did not understand it for what it was then, and we are only beginning to understand it now. I date the revolution to the invention of the transistor, although it was not the transistor itself that made the difference, but what it symbolized. The basics for the transistor, unlike the vacuum tube, came from scientific-mathematical physics. The vacuum tube came from engineering laboratory experimentation.

The transistor was so small that it no longer had a physical-sensory reality—just the mystical belief that there was something in the little blop at the ends of the wire. Have you ever seen an emitter? Maybe it exists, maybe it doesn't. Nevertheless, engineers could gain an understanding by playing with it, although it was more difficult and more complex than the vacuum tube. This was only the beginning of the revolution. By the 1970s, manufacturers were placing hundreds of transistors into a single blob. This actually made life easier because the new blocks could be described in much simpler terms, e.g. an amplifier. The ease of use made the last stage in the revolution. A designer could make a system with thousands of these new blocks. The final systems could then contain extremely complex functions.

Why is the large system the last

stage in the revolution? Because a human being, regardless of his IQ, could not keep the entire sytem in his head. It was too large. The size is what changed everything. If you have tried playing with a home computer, you may have gotten the sense that it is easy. The first simple program gives a great sense of satisfaction. Now consider the problem of understanding a program made up of 50,000 instructions. One of the reverberation machines I designed has the equivalent of 30,000 actions for each sample of audio. Who can understand the behavior of such a system by playing with it? What is the sensory basis for sitting at a computer console editing in all of the commands?

If you open up a defective piece of equipment with the expectation of repairing it, where do you begin to look when you see 500 ICs, some of which have 48 pins. And each of these pins may have 300 different time-shared functions because each IC contains 40,000 transistors. The fact that the input and output wires may be related to the functionality of reverberation does not change the fact that there is no location in the machine that contains "reverberation." Not only can't *you* repair it, but the manufacturer also has trouble. Many of them replace defective equipment and throw away the returned unit. One computer manufacturer throws away $50,000 back-planes, which contain nothing more than connectors and wires, when they have not been able to do a repair within a week.

Your anxiety in dealing with modern technology is shared by those who are the "experts." This is a hard business because the basis for the technology is not physical-sensory; it is intellectual and inferential.

THE LANGUAGE

The fact that high technology is sweeping over the audio profession as well as the rest of society means that there must be some way to understand it. After all, if nobody could understand it, we would not have been able to build it in the first place. The key to understanding is language. Compare the following two descriptions of a defective piece of audio equipment:

> "It's broken because it sounds bad when I play organ music. The music sounded better two years ago, if my memory is correct."

> "With low frequency input signals, there is an additional wideband noise modulation which rises with increased input level. It gets better after the equipment warms up and it only appears on the left channel."

In neither case is the quotation from a sophisticated design engineer who has detailed knowledge of the circuitry. But, the second quotation is obviously from somebody who has the concept of noise modulation in his vocabulary. He obviously has the feel and intuition to be able to relate his sensory auditory experience to such phenomenon as tape modulation noise.

Digital audio has its own language in order to represent new phenomena. Unfortunately, I cannot teach the phenomena by demonstration since **db** magazine is made out of paper. If I could show you what granulation noise was by listening, I would. I can, however, give you a feel for the origin by explaining the mechanism which creates it. This has been my goal.

Learning new ideas and concepts is not expected to be easy. For many of you the digital audio series is more than a little difficult. However, it is not important that you understand everything, but that you get a sense for the way in which one can look at an issue. The notion of language was made very graphic to me a few months ago. I had the opportunity to work with a lawyer on a patent case. In the process, I began to get a feel for the language of patent law. The lawyers have three words for "proving." As a simple scientist I had only one word for proof. A lawyer has the concept of "preponderance of proof," which means that there is more evidence for than against; he has the concept of "proof beyond a reasonable doubt," which means that there is no evidence to the contrary; and he has the concept of "strongly proved," which is in between. When a lawyer says that he has proved something, he means that for the particular legal issue he has satisfied the required proof level. Legally true does not mean scientifically true. After getting a feel for the legal language, I could much more easily communicate with him; the experience did not make me a lawyer. Reading my articles will not make you a digital audio design engineer.

The process of teaching the language of digital audio is made more difficult by the fact that the profession is immature. New concepts and issues appear; old concepts are better understood and refined. In the year 1995, it will be much easier to understand digital audio. Not only will each of us be 11 years older, but the profession will also be. I can remember the first time I was presented with a "problem" in a digital audio delay line. There were five us looking at a "distorted" sinewave that appeared very strange. None of us could figure out what created such a strange looking problem. A year later, I understood that I had seen what we now call image-frequencies at the output D/A caused by inadequate anti-image output filtering. We have the language and that allows us to both communicate and to evaluate the issue. It also means that we can fix and re-design it with ease.

In other professions, as well as in audio, we spend a great deal of effort making the language very sophisticated. Often this sophistication is mathematical, but ordinary words can be used with the mathematics. The math is much less ambiguous but often lacks a physical feeling. Nevertheless, the mathematical language is so powerful that large systems are designed on the computer. In a recent project, all of the design work was done by using a mathematical model of the process on a computer. Over $100,000 was spent just on the computer simulation without ever going into the laboratory. When the answers appeared, the designers began to play with the model to get a feel for the issue. "What happens if I change the...? What happens if I increase the X parameter...?"

When all of the results were achieved, the specifications for the equipment were sent to the shop laboratory for fabrication. The laboratory model was then a test of the computer simulation. If it checked out OK, that meant that the model was OK. Otherwise, the new phenomenon was investigated in order to fix the model. The laboratory was not the place for invention and design; it was the final examination. If you pass, you go to the next step; if you fail you go back to the start again.

We thus see that the last stage of the revolution has reduced engineering to that of information manipulation. Computers are pure information. When we make an accurate model of material, we have reduced it to information. High technology is abstract because information has no physical form.

It is only in the last few years that audio engineers are beginning to realize that part of the profession is being subject to this part of the revolution. Before you argue with me, let me say that the artistic part of the profession will never be subject to pure information because it is an emotional sensory experience. Only the technical part of the profession is changing. The distance between the users and the designers is widening and a common language is required to communicate across the gap.

WORDS

If you think that language is obvious, try taking the following examination. Take a piece of paper and define the following terms. Some of them are from digital audio and some are from the analog domain.

1. dynamic range
2. frequency response
3. saturation non-linearity
4. muddy sounding
5. signal-to-noise ratio
6. gain
7. input impedance
8. echo density in reverberation
9. resonance
10. aliasing
11. limit-cycle oscillation in digital filters
12. quantization noise in A/D
13. group delay
14. filter order

To illustrate the complexity of our language, I would like to take two of these terms and define them clearly. The first example will be dynamic range. This is a badly used term which is usually used incorrectly.

Dynamic range is defined as the range of signal level from the largest to the smallest. It is expressed in a ratio of levels such as dB. Dynamic range has nothing to do with noise! Dynamic range refers to the range of useable or interesting signals. Sometimes, the smallest useable signal is covered by noise. A signal can have much less dynamic range than the tape recorder. Popular music may only have 10 dB of range. That does not mean that the noise level is 10 dB under the peak signal. We could record the signal on a tape recorder with a signal-to-noise ratio of 90 dB. At playback we would still have a signal dynamic range of 10 dB. When using our language carefully, we would say that we cannot record a larger dynamic range than the signal-to-noise of the tape recorder. One term refers to the signal, the other to the recording equipment. The difference in concepts becomes more obvious when we describe a signal such as a flute tone. We can record such a tone and play it back successfully, even when it is several dB under the noise level. The useable dynamic range of the tape recorder is greater than its signal-to-noise ratio. Signal-to-noise ratio is related to the maximum peak signal and the wideband noise in the absence of signal. The fact that one can hear a small signal which is smaller than the noise level describes a property of the human auditory system, not of the signal or the tape recorder.

The above discussion is a demonstration of the power of language when used carefully. I now ask you to evaluate the success of the db series on digital audio in terms of your learning the new language. Your comments are welcome! ■

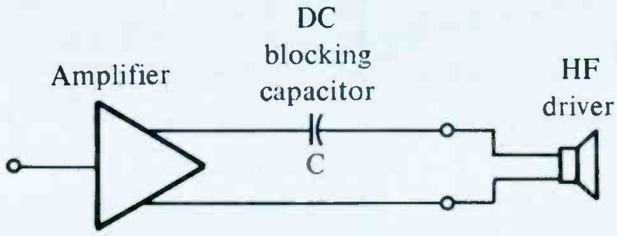
High-Frequency Driver Protection

FAILURE MODES AND POWER RATINGS:

• Compression drivers fail for two basic reasons: thermal overload and displacement overload. Thermal overload is simply burnout of the voice coil caused by excessive power input. Displacement overload may be caused by fairly low power input at frequencies low enough to cause extreme excursion of the diaphragm assembly. Should the diaphragm strike the phasing plug, it will shatter. On other occasions, the voice coil may separate from the diaphragm structure.

A related phenomenon is caused by the aging of some diaphragm materials. Aluminum builds up a stress history, and after a large number of flexures, an aluminum diaphragm may simply fracture, even though no displacement limit has been exceeded. Materials such as beryllium and titanium are much less subject to this phenomenon, and plastic materials do not exhibit the phenomenon at all.

Manufacturers rate their drivers on a statistical basis. A sine wave or a pink noise rating may be easy to duplicate and verify, but program ratings are difficult to assess, due to the great variety of input signals. A driver may be able to handle short high-frequency bursts of power up to ten times the nominal power rating. The short bursts, if they are high



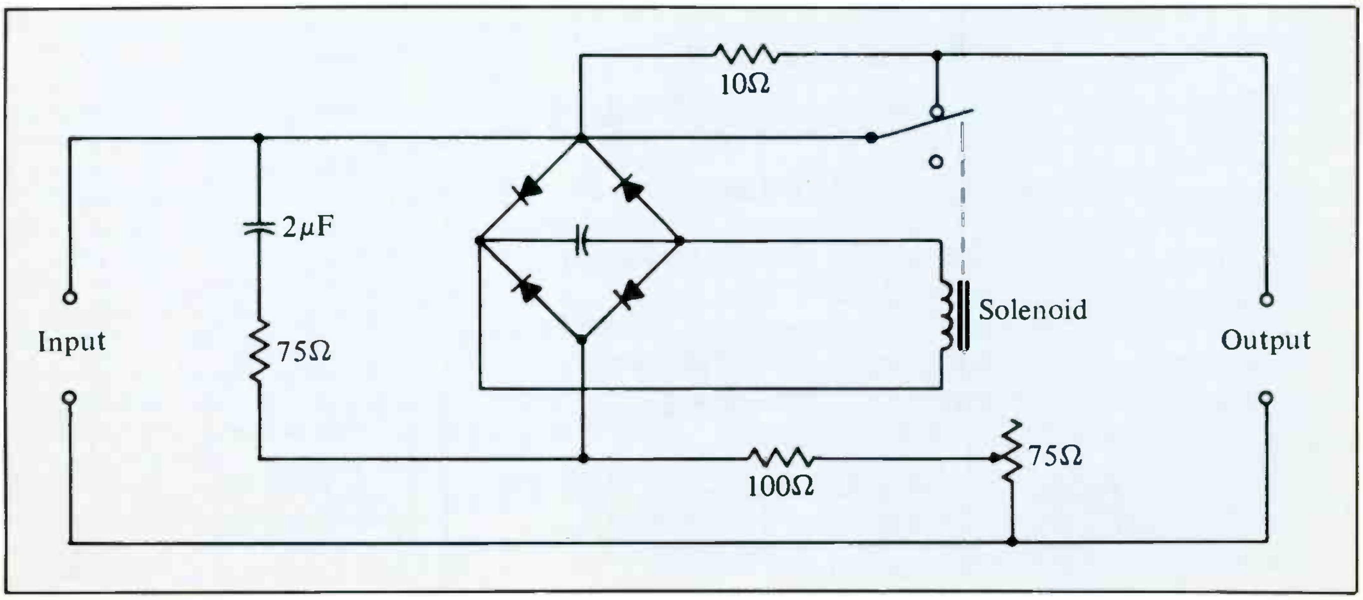
A. USE OF DC BLOCKING CAPACITOR

CAPACITOR VALUE μF	FOR PROTECTION BELOW THESE FREQUENCIES, Hz 16Ω	8Ω	4Ω
72.0	275	550	1100
52.0	400	750	1500
20.0	1000	2000	4000
16.5	1200	2500	5000
13.5	1500	3000	6000
12.0	1700	3500	7000
8.0	2500	5000	10,000
7.0	3000	6000	12,000
6.0	3500	7000	14,000
4.0	5000	10,000	——
3.0	7000	14,000	——

B. SELECTION OF BLOCKING CAPACITOR, C

Figure 1. HF driver protection: use of DC blocking capacitor (A) and selection of blocking capacitor (B).

Figure 2. Schematic of Electro-Voice STR tweeter protection modified for use with compression drivers.



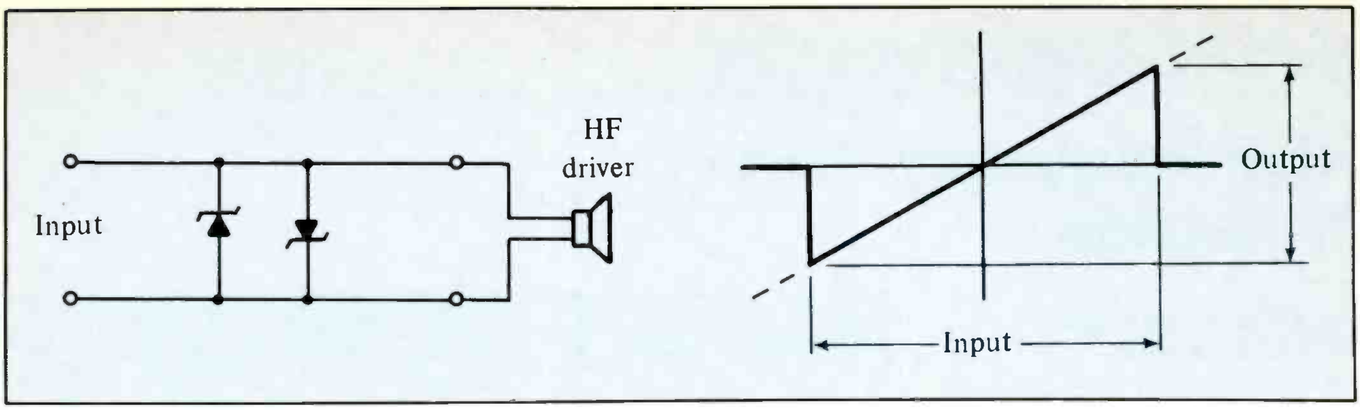


Figure 3. Zener diodes across the inputs of a compression driver.

enough in frequency, do not cause significant diaphragm excursion, and the short duration may not heat the voice coil significantly.

In any normal application, a high-frequency driver is coupled with an amplifier capable of destroying it, and it is the combination of good engineering and operating practice—along with protection circuitry—which may maintain the driver over the years.

SPECIFIC PROTECTION MEASURES:

Blocking capacitors are essential in keeping DC components out of compression drivers. An inadvertant DC surge will cause large displacement of the diaphragm and may wipe out a driver in one pass.

With high-level dividing networks, in-line capacitors are already present, and there is no need for concern. In biamplified systems, the high-frequency amplifier should have an in-line capacitor added, and its value should be chosen according to the data shown in FIGURES 1A and B. The capacitor reactance values are roughly equal to the driver impedance at one-half the nominal crossover frequency.

Not all system designers are in agreement on what kind of protection, if any, should be provided for actual program limiting to avoid over-driving a system. If the system is adequately specified and operated within prescribed limits, then no program limiting may be necessary, other than routine monitoring of levels at the control console. However, where inexperienced operators are at the controls, some form of limiting at the driver may be necessary. FIGURE 2 shows details of a circuit developed by Electro-Voice which rectifies high-signal currents, actuating a relay which attenuates the signal to the high-frequency driver by some predetermined amount. This circuit provides for setting of the threshold of action, and the action is frequency-selective so that potentially damaging lower frequencies are sensed with a lower threshold than are higher frequencies.

The circuit shown in FIGURE 3 may be used for protection of very-high-frequency devices, such as ring radiators, which cross over above 7 or 8 kHz. The Zener diode network placed across the terminals will short out signal overages, once the conduction threshold of the diodes has been exceeded. The circuit produces considerable distortion when in effect, but when limited to the very high frequency range, the distortion may not be perceived as such. ■

JESSE KLAPHOLZ

A Brief History Of Loudspeakers

The essential elements of Loudspeaker design remain basically unaltered since their invention in the early 1900s. In this synopsis, the author brings us up to date on their gradual evolution.

THE FIRST TIME someone said "What did you say?," the need for sound reinforcement was established. How many thousands of years ago this may have been said is anybody's guess. What we do know is that the development of sound amplification started with the speaking trumpet, which dates back to the ancient theatres of Greece.

Modern electro-acoustics' birth is marked by the invention of the telephone by Alexander Graham Bell and Thomas Watson in 1876. Since then, the development of the loudspeaker has been the topic of countless writings, perhaps more so than any other sound system element. In this article we'll take a look at how the loudspeakers we use today developed.

Most loudspeakers in use today have some sort of vibrating diaphragm, although the method by which the diaphragm is set in motion may vary radically. The diaphragm may be one of many shapes, including a cone, a flat surface or piston, a dome, etc.

Where the diaphragm moves as a whole, without breakup, it is analogous to a rigid piston. The characteristics of a circular, rigid piston are well defined, and are essentially derived from the fundamental work of Lord

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

Rayleigh, *The Theory Of Sound*, originally published in 1878. It is Rayleigh's work, and those of his contemporaries, that form the basis of modern sound reproduction.

One of these contemporaries was Heaviside, who introduced the term "impedance." In 1912, the term "motional impedance" was first introduced by A. E. Kennelly and G. W. Pierce. They investigated the impedance of telephone receivers and found that the electrical impedance was dependent upon the coupled mechanical system.

THE INFANCY OF LOUDSPEAKERS

Most of the earliest loudspeakers were based on the moving iron type system. Reproduction was marred by reed resonance, absence of bass, and distortion from the non-linearity in the single-sided operation of the reed. The balanced armature entered the scene in the early '20s, and direct radiating cones began replacing the small horns. Still, there was resonance from the reed, and it was not possible to reduce the stiffness of the motor assembly to yield response below 120 Hz.

The first moving coil invention was the subject of Ernst Wermer's patent, filed by Siemens on December 14, 1877. And on April 27, 1898, Oliver Lodge filed for a patent on the invention of his moving coil loudspeaker. These inventions could not be practically implemented, as there was not sufficient power available in those days to operate moving coil drivers.

Peter Jensen and Edwin Pridham, of the Commercial Wireless and Development Company (which later became the Magnavox Company), patented their invention, the "radio loudspeaker" (a moving coil dynamic driver for a phonograph horn), in January 1913. For the first time, their invention enabled the reproduction of speech and music to be heard over considerable distances.

The Magnavox Company installed and operated the first public address system at the Panama-Pacific Exposition in 1915 at San Francisco. In fact, while the Exposition was going on, there were reports that the crew aboard a navy vessel docked in San Francisco Bay were dancing to "a mysterious music in the air." In 1919, Magnavox provided the PA system for President Wilson's speech at the League of Nations in San Diego. When a Magnavox PA system was used at the Dempsey-Carpentier fight in Jersey City in 1921, the end of the era of the "leather-lunged" fight announcer was marked. Subsequently, Mr. Jensen formed the Jensen Radio Manufacturing Company. As an interesting sidelight, the Western Electric Company produced public address systems between 1918 and 1923 that used horn drivers of Jensen and Pridham's type design.

The mathematical approach to horn design began in 1919, when A. G. Webster showed the superiority of the exponential horn over the conical type. Webster is recognized as the first to publish work on the theory of horns. The complex wave conditions that exist in practical horns were further investigated and refined by C. R. Hanna and J. Slepian at Westinghouse Electric and Manufacturing Company. The results of their work were presented at the Midwinter Convention of the American Institute of Electrical Engineers in February 1924. Hanna and Slepian's publication discussed the throat area, diaphragm chamber volume, final opening or mouth of the horn, shape of the horn, and its damping factor, laying the groundwork for horn design that remained unchanged for 50 years.

The rear radiation of diaphragms has been used to extend the low-frequency response of both microphones and loudspeakers. In 1919, Louis Steinberger designed a telephone receiver that had four small tubes which connected the space behind the diaphragm to the air chamber in the ear cap. The bass reflex principles were first revealed in a patent issued in 1932 to A.L. Thuras, describing the method of increasing the low-frequency response in both a loudspeaker and a dynamic microphone.

Harry L. Duncan was issued a patent in 1923 for constructing a paper loudspeaker cone and its corrugated rim in one piece; Duncan's loudspeaker became the standard for cone type direct radiation loudspeakers. Norman H. Ricker filed a patent in 1922 which was granted in 1932, on a loudspeaker consisting of two large wide angle cones cemented base to base and driven by a moving armature at the center. The Western Electric Company developed and marketed a device of this type, called the 540-AW loudspeaker telephone. It was popular as a broadcast monitor and home receiver loudspeaker from about 1924-28.

Early in the 1920s, Chester W. Rice and Edward W. Kellogg, at the research laboratories of General Electric Company, set out to design an improved loudspeaker. Their first step was to build a 1-watt power amplifier, eliminating the need for relying on resonances and horn loading to maintain adequate output. The result of their work was patented in March 1924 and was published in 1925. During the following year their loudspeaker hit the market as the Radiola Model 104, complete with built-in amplifier, at $250. Thus, Rice and Kellogg are generally regarded as the inventors of the moving coil loudspeaker.

The early moving coil loudspeakers used high resistance voice coils, leather suspensions, and de-powered electromagnets. Within a few years, the moving coil design had made most of its rivals obsolete, and it is still the most widely used type of loudspeaker in professional audio applications today.

HERE COME THE TALKIES

While the early days of the recording and broadcast industries provided the initial spark for producing loudspeakers, it was in the late '20s that the movie industry's quest for "talkies" provided the drive for high-fidelity/high-powered loudspeaker systems. E. C. Wente and A. L. Thuras, at Bell Telephone Laboratories, developed a compression driver for horn-type loudspeakers, the 555-W, that had a continuous 30-watt input capacity, as contrasted with the 5-watt capacity of earlier designs. They further claimed an efficiency increase of 50 percent over other loudspeakers, making their loudspeaker capable of producing 250 to 300 times the sound output of anything previously available.

The development of high powered loudspeakers at Western Electric made possible the "talking motion picture" industry, whose beginning was marked by the historic debut of Warner Brothers' Vitaphone motion picture production, *Don Juan*, in 1926. The work at Western Electric, continuing on into the mid-'30s, created a timeless legacy. These efforts resulted in the 1933 Bell Laboratories' historical demonstration of three-channel sterephonic long-line transmission between Philadelphia and Washington, D.C.

The two-way horn-loaded loudspeakers used for this study, developed by Wente and Thuras, were designed to produce maximum sensitivity over a controlled solid radiation angle with limited signal power input. The power amplifiers were of the single stage type, employing four Western Electric No. 242A vacuum tubes, and were capable of supplying 120 watts RMS. Typically these systems produced sound-pressure levels (re. 1 watt, 1 meter) of about 105 to 107 dB. In the midrange, the directivity index was about 7 to 8, and the corresponding conversion efficiency was about 15 to 25 percent. These types of loudspeakers were large; the low-frequency cabinets were around 11½ feet high, 8 feet wide, and 3½ feet deep. The same kinds of loudspeakers are still in use today in cinema and large scale sound reinforcement applications.

John K. Hilliard of Metro-Goldwyn-Mayer Studios, in 1935, advanced the work of Wente and Thuras by providing a design of theater loudspeaker systems that could be mass-produced and easily adapted to any theater. A selectable configuration of multicellular horns allowed for the simple variation of included angles; large area dynamic low-frequency loudspeaker cones in large horns with a usable response to 50 Hz were used. These low-frequency enclosures had "wings" extending the baffle area, simulating half-space conditions. Among those who worked along with Hilliard in this project were James B. Lansing, Robert L. Stephens, and Harry R. Kimball, and consultants such as Dr. Blackburn and Harry F. Olson. The Lansing 284E high-frequency driver used in this system was substantially the same as many drivers being produced today.

Phase response and arrival times of individual loudspeaker components were of concern to the engineers in the early days. Kellogg, as early as 1924, talked about the propagation time of sound waves travelling through loudspeaker horns, the frequency dependent phase-shift at various points along the horn, and the phase-shift of steady-state tones as contrasted to impulsive sounds. Hilliard and the MGM team were also concerned with the alignment of the loudspeaker system and its phase-response. They were of the opinion that steady-state sounds, such as "sustained music passages," were more tolerant of the effects of misalignment than sounds which "are of the nature of short pulses." A demonstration of loudspeaker alignment was the now famous playback of the film clip of Eleanor Powell tap dancing, as reported by Hilliard in 1936:

> *When this was reproduced it was found that a system with a very small time delay gave a naturalness of reproduction, but systems that had an appreciable delay produced the scene with far less realism. In fact, the sound did not appear to come from the screen, and, in addition, the tap was fuzzy in character with a decided echo.*
>
> *"This effect sounds somewhat like that of transient distortion due to the use of a filter with too-sharp cutoff, but it is actually more analogous to the echo effect often observed on long lines and with certain types of phase distortion networks."*

The loudspeaker system designed by Hilliard and the MGM team won a technical award by the academy in 1936. It was these pioneers in motion picture sound engineering who can be credited with starting the West Coast loudspeaker industry and developing the professional loudspeaker products that we use today.

As we can see, through the 1920s and 1930s, Bell Labs/Western Electric group were the leaders in research and new technology in the audio field. The actual manufacturing of specialized products was done by Electrical Research Products, Inc., a division of the Western Electric Company. One of these specialized areas of E.R.P.I. was the design and manufacture of high quality sound recording and playback equipment.

ALTEC TAKES THE LEAD

Leaving the '30s for a moment, all of us have been affected by the recent breakup of AT&T. However, this is not the first time this has happened. In 1938, by consent decree, the federal government forced Western Electric to divest itself of E.R.P.I. and other divisions. At the time, Western Electric, a major supplier of communications equipment for the military, was falling behind on its orders to the government and needed to concentrate its energies and resources in those areas.

Consequently, a group of E.R.P.I. engineers acquired the assets of their division and continued with the manufacturing of Western Electric sound products. The name of the company? All Technical Products, which was shortened to Altec a year later, probably marking the beginning of "acronyms for audio." The engineers at Altec were now able to focus on the new emerging studio market.

John Hilliard, then at Altec, headed up a project to design a no-holds-barred studio monitor. On October 20, 1943, at the Hollywood Technical Conference of the Society of Motion Picture and Television Engineers, James B. Lansing, one of Hilliard's team at Altec, presented the Altec 604 Duplex loudspeaker. Although there had been previous coaxial loudspeakers by Jensen, Altec, and the Iconic loudspeaker designed by Lansing, it was the 604 that fulfilled all the necessary requirements of a monitor loudspeaker. The 604 quickly took over the market and became the industry standard through the mid-'60s.

No discussion of the development of loudspeakers would be complete without the mention of Harry F. Olson. Olson published information in 1931 on a directional baffle or large-throat horn-loudspeaker for use in sound motion picture theaters, sound reinforcement systems, and Naval applications. During the same period Olson developed the first ribbon microphone, the RCA-44. The RCA-44 became known as a velocity microphone, since the response corresponded to the particle velocity in the sound wave. Olson's RCA-77A unidirectional microphone was introduced in 1933; the 77A was also a ribbon microphone with both velocity and pressure transducers. Both of these microphones are still in great demand today by practically everyone in the recording business.

Olson joined RCA in 1928, and in 1934 he became the Director of the Acoustical and Electromechanical Laboratory. In 1935, he patented the use of passive-radiators in direct-radiator loudspeaker systems. Although commercial use of his patent took years, its physical performance was first described by Olson in his paper "Recent Developments in Direct-Radiator High-Fidelity Loudspeakers" in October 1954, in the JAES.

Olson was responsible for much of the refinement in loudspeaker design and manufacturing technologies through the 1960s. He received more than 80 U.S. patents and authored over 100 papers, articles, and books. In fact, his book, *Acoustical Engineering*, has remained to this day an integral and primary source for loudspeaker design information. I have no reservations in calling it "The Bible of Loudspeaker Design."

Remember Rice and Kellogg at General Electric in the 1920s? While Olson was at RCA, Paul W. Klipsch started his career in the testing department at General Electric in 1926. Klipsch has written many papers and articles about electroacoustics and holds many patents in the fields of geophysics, acoustics, and firearms. He has developed testing methods which include devices for measuring distortion, voice-coil impedance, frequency response, a logarithmic converter to produce a DC output response in decibels of the largest output of a plurality of microphones, and other specialized tools in the field of audio.

Paul Klipsch dedicated much of his work to the investigation of horn theory and application. Based upon the earlier works of Kellogg and Olson, he developed and patented a folded-corner-horn for the reproduction of low frequencies in 1941. Klipsch was also responsible for providing us with much information about loudspeaker distortion. His writings about distortion expanded earlier works on the subject and clearly established differences between amplitude and intermodulation distortion in loudspeakers.

AND ON TO TODAY...

In the '50s, the acoustical lens/horn/compression driver began showing up as a key element of many hi-fi loudspeaker systems. It also became part of the industry standard stage monitor design first implemented by Greg Lucens and Richard Feld in the late '60s.

These lenses were developed at Bell Telephone Laboratories by W. E. Kock and F. K. Harvey, through acoustical applications of the microwave transmission research they were conducting at the time. The work of Kock and Harvey was published in the September 1949 Journal of the ASA, titled, "Refracting Sound Waves." Their paper discussed various acoustical converging and diverging lenses, as well as prisms, and to this day remains the basis for acoustical lens design.

The mid-1970s brought us some new loudspeaker design technology: constant-directivity horns. In 1975, Don B. Keele at Electro-Voice designed a series of horns called the HR series, better known as the "white horns." His work expanded and completed the earlier work and designs of John Gilliom and Ray Newman at Electro-Voice. The E-V ST350A tweeter was probably the first constant-directivity device.

The next step in the development of constant directivity horns was the Manta-Ray horns by Clifford A. Henricksen and Mark S. Ureda at Altec, in 1978. Henricksen and Ureda found that if good directivity control was to be achieved both horizontally and vertically, the mouth size had to be square and relatively large in relation to the low-frequency cutoff point. The Manta-Rays solved the problems of midrange narrowing and loss of "peripheral acoustic power" outside the desired included angle (which Henricksen and Ureda refer to as "waist-banding").

To properly implement all of these features, Henricksen and Ureda had to literally "throw the book out the window," analyze prior designs, and develop new mathematical models. The Manta-Ray horns have no conical, exponential, or hyperbolic expansions, nor do they have any radial surfaces, and their design did not take into account the low frequency response as a paramount element. Horns are usually designed from the throat outwards. However, since the Manta-Ray design started with a given mouth size, it was designed from the mouth inwards—backwards!

The latest type of constant-directivity horns was designed by Don Keele, this time at JBL, called the Bi-Radials. Keele developed some new formulas for the Bi-Radial design, and the equation he used to determine the flare rate of the horn sides is the most outstanding one. This equation, according to Keele, gives a smooth response both along the horizontal and vertical dimension and also off-axis between these two planes. The line, determined by the equation, is rotated about a point on two sides; hence the name Bi-Radial.

Loudspeakers in use today have not changed much from those designed in the '20s and '30s. Better magnet structures, higher temperature adhesives, and materials have advanced those older designs to handle more power with more acoustical output in today's applications.

As we have seen, the loudspeaker industry had its roots in supplying sound for the motion picture industry. The technology used today was provided to us principally by Bell Labs, RCA, General Electric, and Westinghouse (with the additional help of "trickle-down" technology of the aerospace, computer, and defense fields).

The pioneers then, and the engineers now, based their work on the requirements of cinema and what we can refer to as conventional entertainment. With entertainment evolving at the current digital/computer-influenced rate, we are now faced with radically changing ingredients of the design recipe. ■

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JOHN EARGLE

Monitor Loudspeakers

Here, the author reviews the history of the monitor loudspeaker and clarifies the criteria for identifying one.

Figure 1. The Iconic monitor system. Built by Lansing Mfg. Co. during the '30s. Note power supply on top for powering magnetic field coils.

FROM THE EARLY DAYS of recording, broadcasting, and sound-on-film, monitor loudspeakers have been an important element in the creative chain. Engineers may not always agree on just what a "monitor loudspeaker" is, or should be, and it is certainly true that in recent years the term has been used quite loosely.

Over the last half century, it is likely that there have been no more than two dozen loudspeaker models produced by a handful of manufacturers that have been accorded the status of "monitor loudspeaker" by the industry.

In this overview of monitors, we will first discuss their historical evolution, and then cover the specific technical aspects of performance which define them.

A BRIEF HISTORY

Sound-on-film and electrical disc recording began their commercial ascent within a few years of each other. In this country, the dominant companies in these efforts were Western Electric and RCA. The work of Lansing, Hilliard, and Shearer at MGM was equally important, and Lansing's Iconic system, a two-way design, was

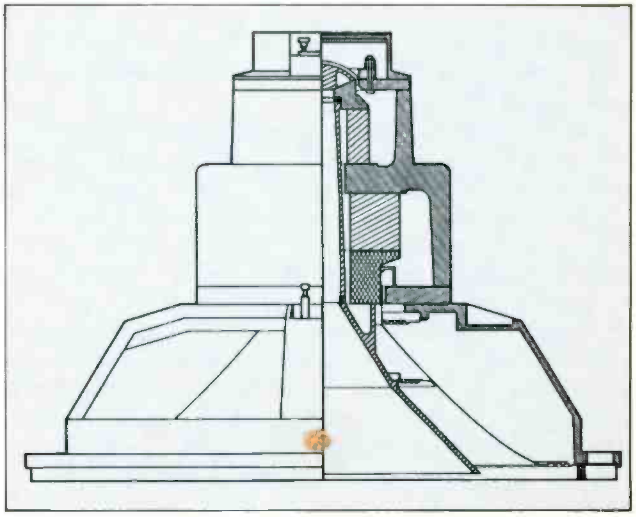

Figure 2. Altec Model 604-8H coaxial loudspeaker. Sensitivity is 101 dB, 1 watt at 1 meter. Input power rating is 150 watts (Altec data).

John Eargle is a consultant for JBL and is the Technical Advisor of db Magazine.

probably the first loudspeaker that could be referred to as a monitor by the industry (see FIGURE 1). The Iconic used a 15-inch low-frequency (LF) driver that crossed over at about 800 Hz to a small high-frequency (HF) multicellular horn driven by a 1¾-inch diaphragm driver.

The Iconic was essentially a scaled-down version of the larger two-way screen system on which Lansing had collaborated with Shearer and Hilliard. It was intended for use in small screening rooms and recording control rooms. The record industry adopted it as well for general use.

The next significant development was the Altec Lansing 604, that hardy perennial that has endured to this day. In 1941, the Altec Service Company absorbed the Lansing Manufacturing Company, and Jim Lansing

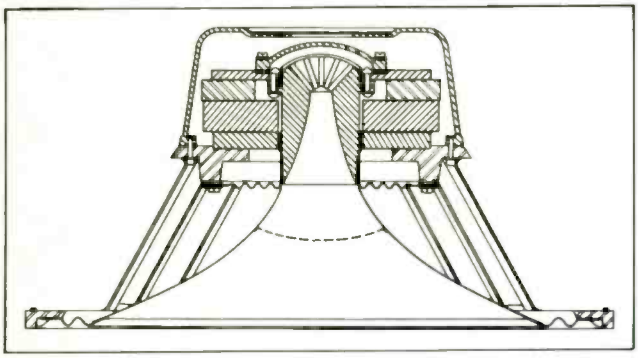

Figure 3. Tannoy 15-inch Dual Concentric loudspeaker. Sensitivity is 94 dB, 1 watt at 1 meter. Input power rating is 120 watts (Tannoy data).

was made vice president of engineering. During the mid-'40s, he developed the 604 Duplex loudspeaker and conceived it as something of a scaled down Iconic intended for use at short distances in small recording environments. FIGURE 2 shows a cutaway line drawing of the current model 604.

During the '40s and '50s, Altec had the monitor business pretty much to themselves, and the addition of the A-7 to their line defined the models of choice for both control room and studio monitoring applications.

In England, Tannoy, under the direction of Guy R. Fountain, introduced the "Dual-Concentric" loudspeaker, a design similar to the 604, and it was to become the standard for recording monitoring in that country. FIGURE 3 shows a cutaway line drawing of the current 15-inch Dual-Concentric model.

In Hollywood, things were about to change. The Capitol Records pressing plant on Fletcher Drive was within a few blocks of the JBL factory on Casitas Avenue, and the late Ed Uecke, head of engineering for Capitol, wanted to develop a monitor system with extended HF and LF response. The proximity of his office to JBL made them a logical choice. At that time (the mid-'60s), Bart Locanthi, presently with pioneer North America, was vice president of JBL. He worked with Capitol on the development of what was to become known at the JBL 4320 system, and EMI, Capitol's parent company, adopted the system for use in their studios around the world. Thus began JBL's rise in the professional monitor field.

During the '70s, Electro-Voice made a significant entry into the monitor field with their Sentry III model,

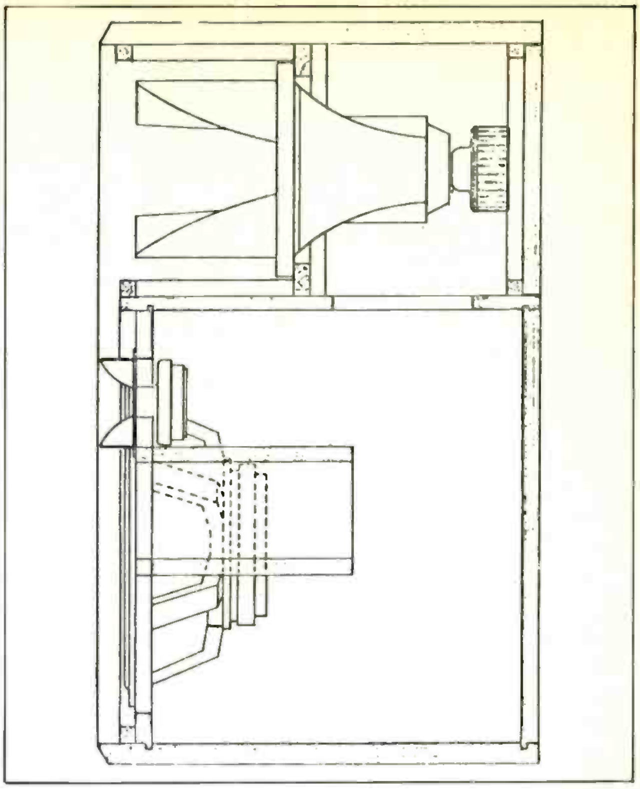

Figure 4. Electro-Voice Sentry III monitor system. Sensitivity is 97 dB, 1 watt at 1 meter (E-V data).

shown in FIGURE 4. Their market penetration into broadcasting, by way of their extensive microphone line, helped them become a major supplier of smaller monitors to that industry.

By the mid-'70s, the 604 was ready for resurrection—two of them, in fact. The first of these was by way of the Mastering Lab. They introduced a dividing network for the 604 that became quite popular. Of more significance perhaps was the work of Bill Putnam and UREI in establishing their "Time-Align"™ versions of the 604. Working with Ed Long, they introduced three models of

Figure 5. UREI Models 811 (top), 813 (left), and 815 (right) monitor systems. Sensitivity of the 811 is 99 dB, 1 watt at 1 meter. Input power rating is 150 watts (UREI data).

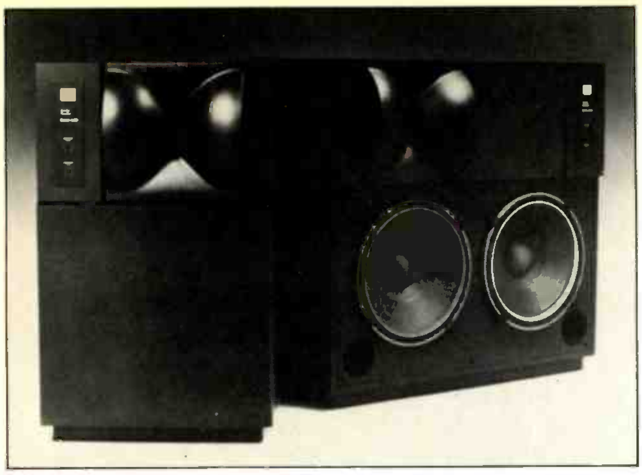

Figure 6. JBL Models 4430 (left) and 4435 (right) monitor systems. Sensitivity of the 4430 is 93 dB, 1 watt at 1 meter. Input power rating is 300 watts (JBL data).

monitors designed around the 604, using dividing networks that included delay sections in the LF portion to bring the two sections of the system into time coherency. During the late '70s and early '80s, the UREI monitors achieved a very high status in the industry, proving all over again that recording engineers have always loved coaxial designs. FIGURE 5 shows one of the UREI designs.

Partly as a response to the success of the UREI monitors, JBL rethought its monitor philosophy and introduced the 4400-series of Bi-Radial designs, as shown in FIGURE 6. These designs were based on the notion of flat power response as well as the general preference of pop recording engineers for two-way systems.

In March of 1983, Harman International, the parent company of JBL, acquired UREI, making Harman the largest supplier of large studio monitors in the business.

The popularity of smaller, bookshelf monitors in the last 10 years has reflected the growth of the so-called semi-pro studio, and there are many suppliers for that market. One of the most ubiquitous of the mini-monitors is the small single-cone Auratone, a pair of which may be seen perched on the edges of the console in almost any pop or rock studio.

With this brief historical survey of monitors out of the way, we now move on to a discussion of the particular characteristics which define monitor loudspeaker systems.

TECHNICAL ATTRIBUTES

Reliability and accuracy are the prime requirements. The requirements for reliability are relatively easy to define, while those of accuracy are more difficult to assess.

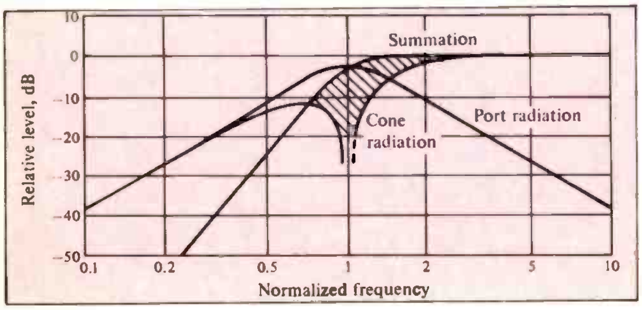

Figure 7. Relative acoustical contribution of port and cone in a ported system. At resonance, cone motion and distortion are minimized.

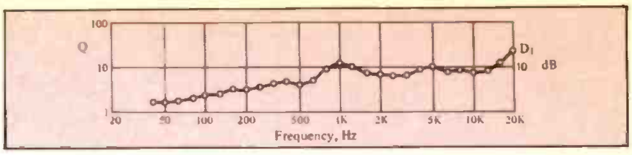

Figure 8. Directivity Index for a two-way monitor system with 90- by 90-degree HF coverage. LF transducer is 15 inches in diameter.

Reliability: Most of the larger monitors make use of ported LF systems and compression-driver/horn HF systems. These two approaches are likely to draw fire from certain segments of the audiophile community, but there are good reasons why monitor design has gone in those directions.

FIGURE 7 shows how a properly ported system lowers distortion by lessening the amount of excursion a low frequency transducer is called upon to execute. Response in the 30-to-40 Hz range is important, and there are few sealed LF systems, short of multi-driver arrays, that can generate the acoustical output of half the number of drivers in a ported configuration.

The use of horn HF systems has characterized all of the large monitor designs from the very beginning. Horn HF systems are always padded down some 8 to 10 dB relative to the LF system, and thus the HF voice coil is called upon to dissipate relatively little power. Accordingly, they are lightly stressed and they last a long time. They are also tolerant of accidents, such as bursts of HF power, which may be generated when the headgate of a tape recorder is shut during fast rewind.

While cone and dome HF systems can produce smoother response than the horn systems, their absolute power limits are more likely to be well within the normal acoustical monitoring range of the typical pop-rock studio. However, for purely classical monitoring, or the monitoring of jazz recordings at moderate levels, there are many cone-dome systems that hold their own very well. B&W of England, for example, have made significant inroads into classical monitoring with their three-way systems using all direct radiators.

In construction, we are likely to find the transducers in monitor systems made of cast and machined parts instead of the stamped pieces common in most consumer systems. The reasons here are simply that casting and machining help to maintain the tight tolerances which are often an essential part of high-efficiency magnet structures. Enclosures will usually be made of one-inch thick plywood or particle board in order to hold down spurious resonances.

Flat Response: While this was not always true, today's monitors are reasonably flat in output. There are two aspects to this: flat on-axis response is important, inasmuch as the first arrival sound tells the mixing engineer a great deal about what he is mixing. In addition, the off-axis response, representing the total power output of the system, should be smooth. When this is the case, the reflected sound field in the monitoring environment will be free of peaks and dips, and decisions regarding overall balance will be more accurate.

If the on-axis frequency response of a monitor loudspeaker is flat, the directivity index (DI) of the system becomes an inverse measure of its power response. DI is a measure of how directive the system is on axis, as compared with the same acoustical power radiated in all

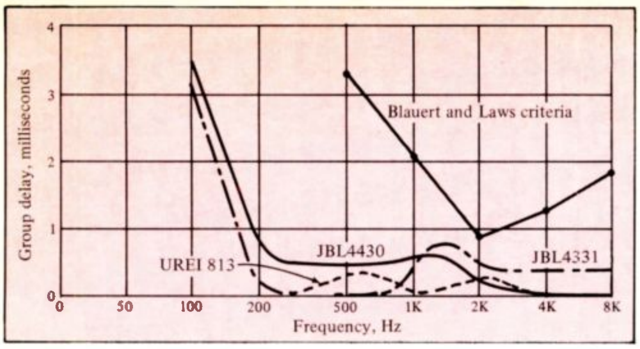

Figure 9. Group delay characteristics of typical monitor loudspeakers.

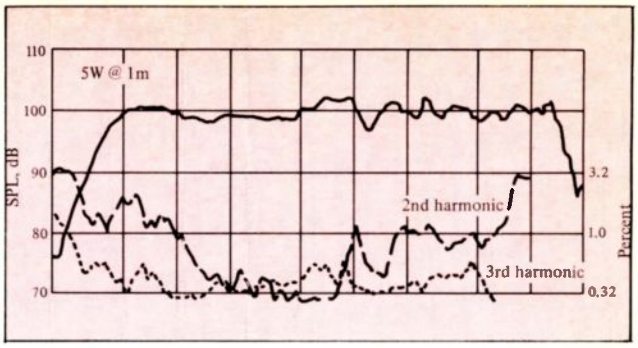

Figure 10. THD for a two-way monitor system with 5 watts sine wave input.

directions. An example will help the reader to understand this.

FIGURE 8 shows the DI of a two-way monitor design. Note that the DI rises from about 3 dB at low frequencies up to a value of about 10 dB at 1 kHz. From that point upward, the DI remains constant within ±1 dB, signifying that the radiation angle of the HF horn (in this case, a constant coverage design) remains fixed up to about 12 kHz. The goal of such a design is to avoid hot spots or holes in the reflected sound field of the listening space.

Time Coherence: Early workers in sound motion pictures carefully adjusted the fore-aft positioning of the HF part of their systems for transient accuracy. The adjustment was made while playing a recording of clicks or transients, such as those produced by the soundtrack of a tap dancer. The HF carriage was moved in and out until the sound clearly fused into one precise click.

In much later times, Blauert and Laws[1] have determined the limits of group delay, or delay error, as a function of frequency, and their criterion is shown in the graph of FIGURE 9. We have plotted on this graph the group delay errors of three current monitors. The UREI 813 clearly shows the effect of precise aligning of the two sections of the system, and the group delay error in the midband is virtually zero. The slight perturbations in the 500 and 2000 Hz range are associated with the amplitude response of the system. The JBL 4331 three-way horn system exhibits a swing in delay in the 1 to 2 kHz range, and the JBL 4430 two-way system exhibits a smooth shelving action between the HF and LF parts of the system. All three systems show the expected rise in group delay at lower frequencies caused by system roll-off at low frequencies.

Note that all three of these systems fall within the audible limits established by Blauert and Laws. As a rule, bookshelf systems have negligible errors, while those larger monitors having long midrange horns may exhibit the greatest errors of all.

Distortion and Power Compression: For pop and rock recording, monitors are often called upon to produce quite high levels for long periods of time. Levels at the control console often get as high as 110 or 115 dB SPL, and there are few systems designed for the home that can deliver such levels.

The usual measure of distortion is Total Harmonic Distortion (THD), and it represents the percentage of output power at frequencies other than the single input frequency to the system. FIGURE 10 shows typical THD measurements for a large two-way design with a compression driver HF section. For making meaningful comparisons, THD should be related to a reference acoustical output level rather than to a reference electrical power input level.

Power compression results from the increase of voice coil resistance as a result of heating. In some ways, this increase in resistance acts to protect the transducer, but its effect on a program can be quite noticeable. Over the input range from 1 watt to 100 watts, a large monitor may not exhibit more than, say 1 or 1.5 dB of compression. High sensitivity devices and large voice coils are the best insurance against this phenomenon. FIGURE 11 shows the power compression of a typical large monitor with a single 15-inch LF transducer with a 4-inch voice coil.

Stereophonic Imaging: Just as with audiophile home systems, there is a growing concern in the studio for precise stereo imaging. The important details here are accurate matching of components in stereo pairs of loudspeakers as well as mirror imaging of the two loudspeakers. In control rooms, the mirror imaging is often extended to the boundary conditions themselves, and every attempt should be made to design control rooms with complete bilateral symmetry with regard to reflective and absorptive treatment.

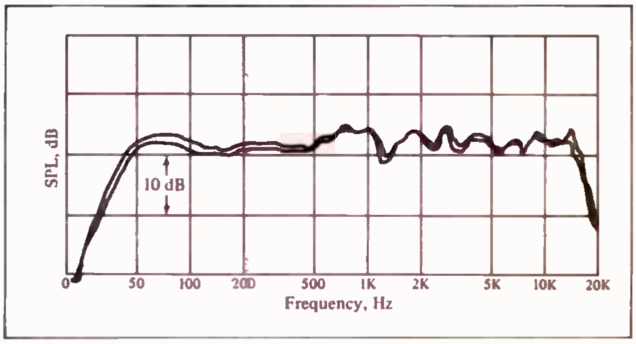

Figure 11. Power compression at three input levels. LF transducer is 15 inches in diameter.

Loudspeaker design is a careful balancing of attributes. It is impossible to have all the good things and none of the bad; tradeoffs are demanded. If we could all agree on the ordering of the attributes, then there would be only one loudspeaker model in each price range! But such will never be the case; there will always be different monitors to choose from, even for the same job.

Remember, if it isn't used by some segment of the professional sound industry for some purpose, it really isn't a monitor. ■

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Amplifier-Loudspeaker Interfacing

A. B. KRUEGER

Amplifier-loudspeaker mismatch? This often overlooked problem may be causing listeners to "miss out" on some pleasurable information.

THE EXISTENCE of amplifiers of various power ratings begs the question, "How much amplifier power is necessary?" Unfortunately, the question is usually answered with another question: "How loud do you want to play the music?" Answers to that question are inherently subjective, leaving the question without a general answer.

Subjectivity can be bypassed by assuming that the decision as to how loud one desires to hear the sound is implicit in the choice of loudspeakers. Since loudspeakers are usually thought to be the system's weakest link, the remainder of the components need only be strong enough to "break the weakest link." As a practical matter, it is undesirable to break the loudspeaker, and a well designed system will save it from damage using a fuse or other form of protection.

Thinking that the loudspeaker is the weakest link, one might select a power amplifier with just enough power to trip the loudspeaker protective system without clipping. An exception to this is when highest possible sound quality is not the goal; amplifier clipping can be an effective means of protecting loudspeakers when all that is desired is loud sound. Another exception is when the loudspeaker is capable of generating very large amounts of acoustical output without damage to itself. The estimated amplifier size in this case is one capable of driving the loudspeakers to levels verging on severe listener discomfort or ear damage, without clipping. Objective criteria for severe discomfort and ear damage exist, so subjectivity is avoided.

Loudspeakers, and therefore effective loudspeaker protection systems, have the ability to handle extremely large power levels for short periods, but can only handle much smaller amounts of power for extended periods. Since most music is constantly varying in level, the ability to handle large amounts of power for brief periods is exercised in normal operation. The ratio of peak-to-average power levels in music is at least 10 dB, so the amplifier would then need to have at least 10 times more power than it would take to trip the continuous power capacity of the loudspeaker protective circuit (if it is to trip it without clipping) when reproducing music.

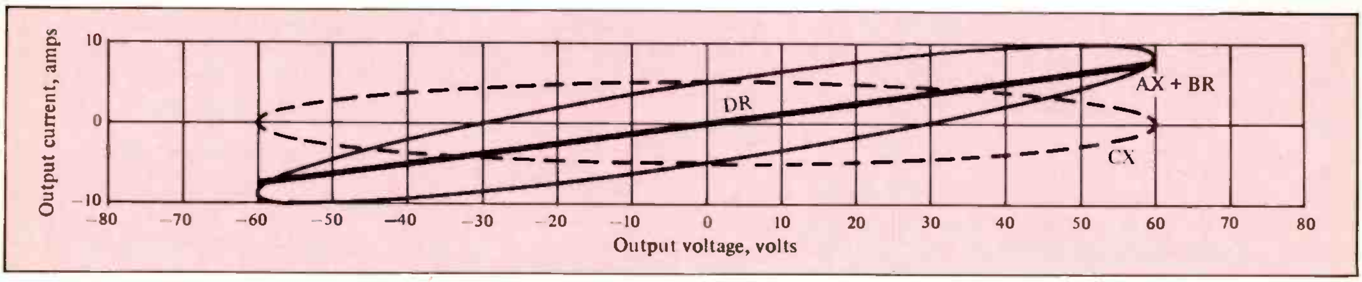

Figure 1. Load lines map the voltages and currents for various kinds of loads. The solid line is a typical mixture of resistance and reactance seen with loudspeakers. The dashed line is a "pure" reactance. The heavy solid line is for a "pure" resistance.

FINDING THE PERFECT MATCH

To evaluate the practical merit of these ideas about selecting power amplifiers, a loudspeaker and amplifier are matched. The loudspeaker has a nominal impedance of 4 ohms, and measurements indicate that this is its approximate impedance over much of the audible range. It is fused with 3 amp-type 3AG fuses which are neither slow-blow nor quick-blow. Several years of playing at loud levels in a medium-size room with no failures indicates that the fusing is adequate to protect the loudspeaker. 3 amperes into a 4-ohm load is 36 watts of power, so a power amplifier capable of 360 watts into 4-ohm resistive loads is used, giving about 10 dB of peak-to-average headroom. However, in use, the amplifier can be driven into audible clipping on popular music without blowing fuses.

Some music has a peak-to-average ratio of more than 10 dB, which helps explain some of the clipping. However, the ear can tolerate some clipping, and audible distortion was not expected. An oscilloscope used to monitor the output of the amplifier shows that the amplifier's output voltage at clipping is sometimes about half of its capabilities into a resistive load, which means that at times, its actual maximum output power is one quarter of its ratings.

Amplifier clipping at about one quarter of its rated power output indicates that despite a reasonable power rating, it is not capable of fully exercising the loudspeakers. An ideal match does not exist. This contradicts the idea that the loudspeaker was the weakest link; in this case the amplifier is the weakest link. One might question

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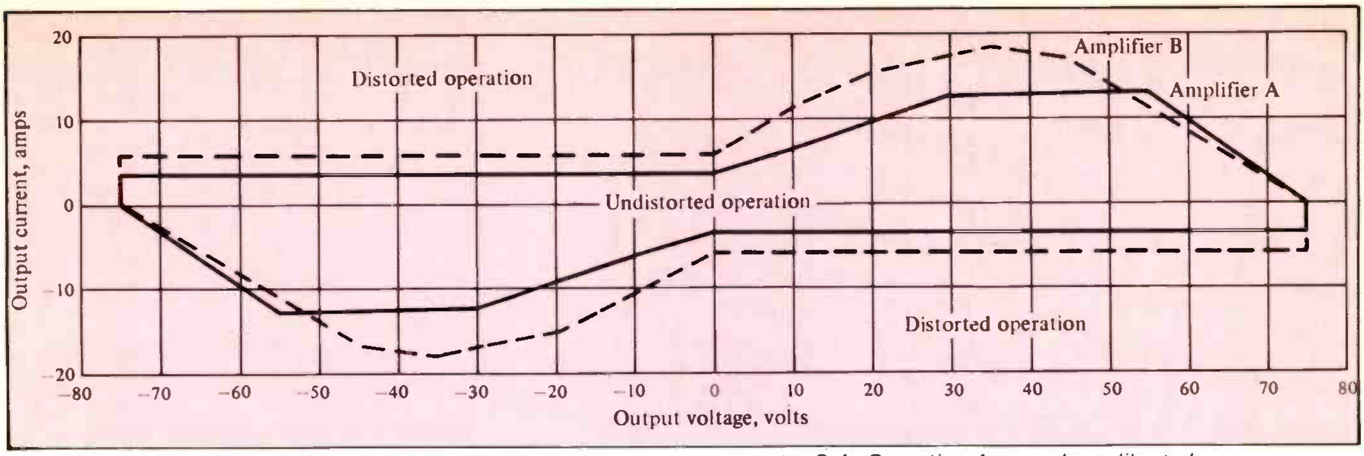

Figure 2. Amplifier capabilities mapped in the voltage-current plane. The line with a single break represents the capabilities of an older high-powered class AB solid state power amplifier. The line with two breaks is the same amplifier updated with newer output devices with greater Safe Operating Area and recalibrated dissipation limiting circuits. Undistorted operation lies within the lines. Updating the amplifier significantly increased its ability to drive loudspeaker loads despite the fact that performance into resistive loads was unchanged.

the quality of the amplifier. Some persons are very suspect of the ability of amplifiers to drive certain loudspeakers effectively, and may even consider loudspeaker-amplifier interfacing an art.

It is generally accepted that improper loudspeaker-amplifier matching can cause the absence of expected acoustical output, various undesirable noises from the loudspeaker, and unexpected failures of amplifiers and loudspeakers. Equipment failure did not occur in the example because of the robustness of the loudspeakers and the conservatism of the amplifier's design, but damage might have happened just as easily. The usual approach to solving this problem is to just try another amplifier, perhaps one a little "beefier." But random application of resources to achieve reliability is craftsmanship, not engineering. If the situation was better understood, equal quality and reliability could be achieved at a lower cost, or better could be had for the same cost.

It is well-known that loudspeakers present a reactive load, and that amplifiers vary in their ability to drive reactive loads. "Load lines" of resistive, purely reactive, and complex loads are shown in FIGURE 1. This chart shows the current that must be delivered to the loads to achieve a given voltage drop.

REACTIVE AND RESISTIVE LOADS

Load lines, as shown in FIGURE 1, are generally elliptical, but shrink to straight lines for resistors. One way that reactive loads differ from resistive loads is that a reactive load can have considerable current flowing through it while the voltage across it is zero. This is because the relationships between voltage and current in a reactive load are based partially on history, while resistors operate on the situation "now." Inductive and capacitive load lines of equal reactance are the same ellipse, but the path traced by the output signal is counterclockwise for inductors, and clockwise for capacitors.

Amplifiers receive additional stress while driving reactive loads because the amplifier must supply significant currents when its output voltage is zero. This creates a situation in which the amplifier is simultaneously providing substantial currents, and has large voltage drops across internal parts. With a resistive load, this does not happen.

Peak currents are only supplied to resistive loads when the amplifier's output voltage is high. This means that the output voltage of the amplifier is close to the voltage of its power supply, and voltage drops across internal parts are minimized. Since the power dissipated by the amplifier is the product of internal voltage drops and the current delivered to the load, reactive loads require amplifiers to dissipate more power than do resistive loads with the same impedance.

Amplifier power dissipation at zero voltage output is not the worst case. It is possible for an amplifier to charge a capacitive load to a high voltage, and then have the program material require that the voltage across the load be reversed. This places twice the voltage of the amplifier power supply across its internal parts. A more esoteric situation with similar results can come about through the interaction of amplifier protective circuits and inductive loads. A reactive load with reasonable-sounding impedance magnitude can cause an amplifier to dissipate more heat than any resistive load including a short. Thus, existing tests based on resistive loads are nowhere near the worst case for amplifier power dissipation. Resistive load tests don't stress amplifiers like some real-world loudspeakers do.

THERMAL SOLUTIONS

Greater power dissipation leads to greater heat build-up. High temperatures lead to reduced reliability. All amplifiers have power dissipation limits, and their operation should be controlled to fall within their limits for reliable operation. This is true for amplifiers built of tubes, bipolar transistors, FET transistors, and is especially critical for amplifiers with 100 or more watts of power output.

The fact that reactive loads can cause much greater heat build-up in amplifiers than comparable resistive loads becomes a sticky economic issue, because handling power and dissipating heat tends to make amplifiers large, heavy and expensive, while consumer demand is for amplifiers that are small, light, and easy to buy. While much has been made of the effect of the "unalterable laws of physics" on loudspeaker design, there are similar effects with amplifiers.

Some form of power dissipation control is included in well-designed equipment. It may take the form of fuses, current-limiting circuits of various degrees of complexity, thermal cutouts, transformers that saturate and limit current, or active devices that decrease their

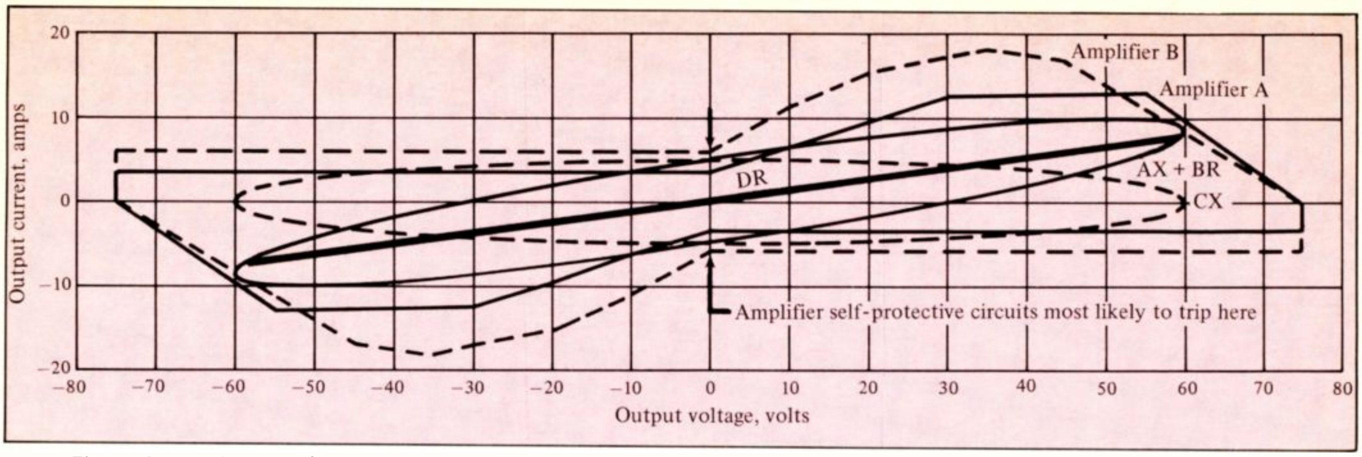

Figure 3. Merging the capabilities of amplifiers with the requirements of different kinds of loads shows that the point where amplifier protective circuits are most likely to be tripped is when output voltage equals zero.

gain when they are stressed. A possible route for ensuring reliable operation, avoiding other forms of protection, is to package the amplifier with a compatible loudspeaker. The packaged environment becomes the protection for the amplifier.

The first step toward an improved understanding of amplifier-loudspeaker matching may be to view amplifier capabilities is the voltage-current plane, as shown in FIGURE 2. This "map" of power amplifier capability can be devised by testing or circuit analysis or both. The two amplifiers shown are identical except for dissipation-limiting circuit trip-point settings.

Various techniques can be used to measure the information in FIGURE 2. Simple capacitive loads are very effective tools for making these measurements because they load the amplifier differently at different frequencies. The test frequency can be changed to get different loading effects, which is easier than actually changing the load. Tone bursts of duration on the order of one second can be used to achieve the high power levels possible with music waveforms, for circuits that would be damaged or blow fuses in continuous sine-wave operation. Monitoring the current and voltage delivered to a reactive load with a random noise signal as program material is also a way to probe the limits of an amplifier.

Many factors can contribute to the limits of an amplifier's operation in the current-voltage plane. Dissipation-limiting circuitry can include reactive elements such as capacitors, making the trip points dependent on recent history, or signal frequency. FETs are temperature sensitive, and while they protect themselves, they will also decrease maximum undistorted output as they get hotter. Since the trip points of bipolar amplifiers are set based on estimates of operating temperature, it might be reasonable to include thermal sensors in their protective circuits. It is common to idealize bipolar transistor protection circuits for capacitive loads operating at frequencies above 10 Hz.

Higher power line voltages increase voltage drops, and lead to higher temperatures. The higher power supply voltages required by FETs can lead to greater heat build up, especially with reactive loads.

REACTIVE DRIVE

FIGURE 8 adds the load lines characteristic of various resistive and reactive linear loads originally shown in FIGURE 1. Closer examination of the demands placed on an amplifier by reactive loads shows that the current output at zero voltage output is a point where common protective circuits are most likely to be triggered. Therefore it makes sense to have an amplifier characteristic, called Reactive Drive. The Reactive Drive (RD) of an amplifier is defined as the maximum undistorted current deliverable to a reactive load when the output voltage is zero, divided by the current deliverable to a resistive load of the given impedance. A new, more complete specification of the output capabilities of an amplifier includes both power output and RD as a function of load impedance. RD is primarily a function of load impedance and power, but can also vary with signal frequency, power line voltage, and temperature. RD for most amplifiers is fairly constant in the audible range for most amplifiers, but there may be variance in measurements with inductive and capacitive loads, if the protective circuits contain time constants.

In many cases, RD can be approximated from existing power output versus load impedance information, once the maximum undistorted output current at zero output voltage has been determined. Data points for common load impedances taken from FIGURE 3 along with RD for the two amplifiers are shown in Table 1.

LOAD IMPEDANCE (OHMS)	POWER (WATTS)	PEAK VALUE OF I_{MAX} (AMPS)	AMPLIFIER A		AMPLIFIER B	
			I (V_{OUT} = 0) (AMPS)	RD	I (V_{OUT} = 0) (AMPS)	RD
4	364.5	13.50	3.6	.267	5.8	.430
8	240.3	7.75	3.6	.465	5.8	.748
16	144.5	4.25	3.6	.847	5.8	1.370

Table 1. RD and power output calculated from FIGURE 3. Maximizing RD makes an amplifier more capable of driving reactive loads.

With a resistive load, the voltage across the load is in phase with the current through it, and when the voltage at the output is zero, the current will also be zero—a desirable condition. When a voltage is applied to a reactive load, the current flowing through the load is not in phase with the voltage. If the current is out of phase with the voltage, then the current will not be zero when the output voltage equals zero—an undesirable condition. The phase shift between the voltage and the current can be calculated from the ratio between the reactance of the load to its impedance. The greater the ratio of reactance

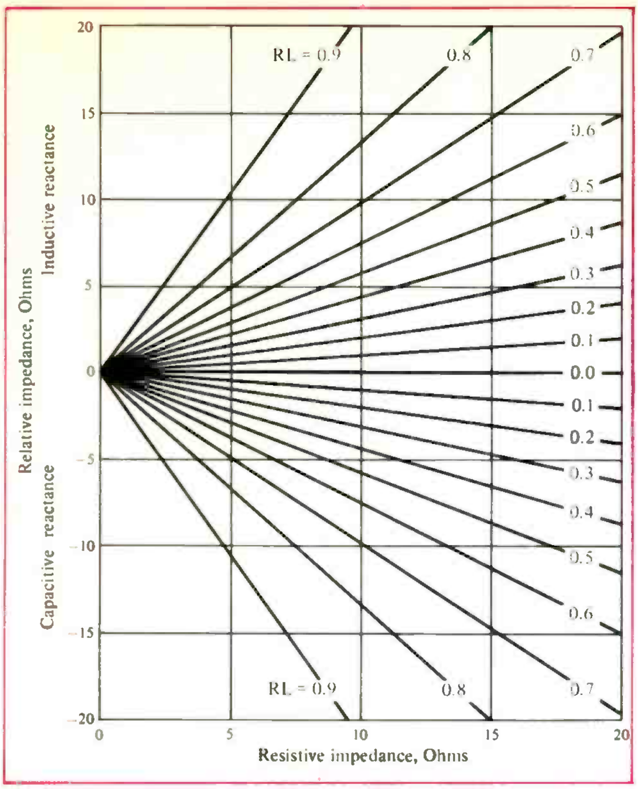

Figure 4. Reactive Loading (RL) is defined so that it is a relative measure of current at output voltage equals zero, required to drive a reactive load to an appreciable output. Quite by coincidence, it plots as straight lines in the plane of complex impedance, used by some to characterize loudspeaker impedance.

to impedance, the greater the phase shift between the current and the voltage. For a resistive load, the ratio of reactance to impedance is zero, while the ratio for a purely reactive load is one. The phase shift between voltage and current in a reactive load puts greater stress on the amplifier than there would be for a resistive load of comparable impedance.

REACTIVE LOADING

The ratio between the reactance and the impedance of the load characterizes the phase shift between the voltage across it and the current flowing through it, and thus the stress the load puts on the amplifier. This ratio is thus defined as the Reactive Loading (RL) of a loudspeaker. It is also the ratio of the current drawn at output voltage equals zero divided by the magnitude of maximum current drawn. RL for loudspeakers is very similar to RD for amplifiers. RL can vary with frequency, power level, and temperature of loudspeakers. Loudspeakers can be very nonlinear, and specifying RL at peak and average power levels can be revealing.

Richard Heyser has been publishing plots of reactance and impedance for loudspeakers in his *Audio* magazine loudspeaker test reports for several years. Points of constant RL plot as straight lines on those plots, as is shown in FIGURE 4. Current-limiting is the amplifier fault being concentrated on, and it is minimized by staying on the X axis (horizontal line) and moving towards the right. This corresponds to the line for RL = 0.0, and maximizing impedance magnitude.

If the RD of an amplifier is less than the RL of a loudspeaker, then the amplifier will not be able to drive the loudspeaker to the rated power output of the amplifier without current-limiting. Plotting RD of amplifiers on a complex impedance chart as shown on FIGURE 5 establishes their ability to drive loudspeakers. This plot was made from TABLE 1. A compass was set to the various load impedances of 4, 8, and 16 ohms by measuring along the X axis, marking an arc through the lines of constant RL. Amplifier RD is interpolated along the arc and marked. Lines showing amplifier RD are drawn through the marks. Since the lines of constant RL are symmetrical about the X axis, the lines of amplifier RD are symmetrical if its protection circuits have no time constants that are significant in the audible range. Above the upper RD curve, or below the lower RD curve, lie regions of loudspeaker impedance that will cause the amplifier to clip prematurely.

A loudspeaker's worst case load on an amplifier may be characterized by its maximum RL, and minimum impedance. RL can often be improved by connecting resistors in series with the loudspeaker, with an obvious loss in efficiency. In some cases, the reduction in RL and increase of impedance caused by a serious resistor will allow undistorted operation at enough greater amplifier power output that efficiency losses will be more than offset, and undistorted loadness will increase.

Minimizing RL and maximizing impedance makes a loudspeaker easier to drive, if efficiency is held constant. FIGURE 6 adds impedance curves of three loudspeakers to FIGURE 5. A diagram like FIGURE 6 can be used to evaluate the match between given loudspeakers and

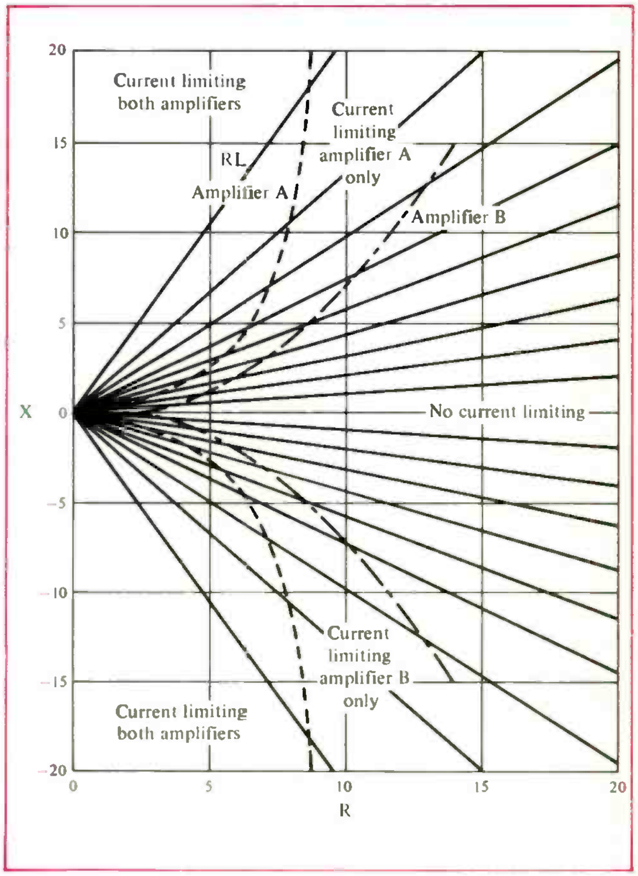

Figure 5. Reactive Drive (RD) is defined as a measure of the maximum available current when the output voltage equals zero, which is required to drive a reactive load to an appreciable output. It is related to load impedance, and can be plotted on the plane of complex impedance. Areas of no current-limiting (desired operational mode) and current-limiting (undesired operational mode) are shown.

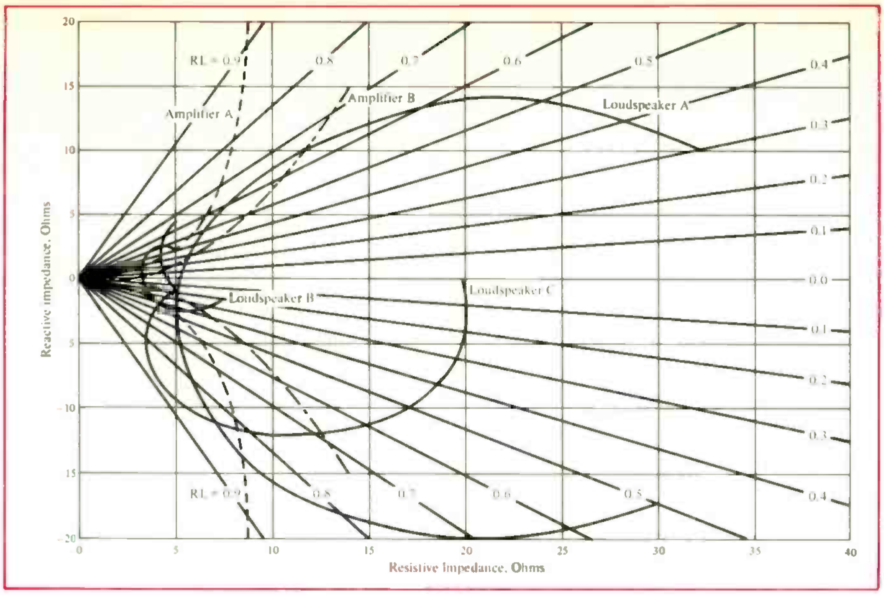

Figure 6. Plotting loudspeaker impedance characteristics onto the complex impedance plane completes the evaluation of amplifier ability to drive loudspeakers without current-limiting. As can be seen, all of the loudspeakers will cause both of the amplifiers to current-limit. The loudspeaker impedance curves are drawn from figures by Richard Heyser published in Audio Magazine *for devices that are more stressful than most.*

amplifiers. Both amplifiers will current-limit with the loudspeakers at some frequencies, though amplifier B is better because of its higher RD.

The goal of matching of RD to RL is to ensure that if amplifier clipping takes place, it is the result of voltage limits, not current-limiting. Current-limiting with reactive loads can be destructive because it usually causes a sudden shift of the output stage to a high impedance state. The inductive load will maintain a constant current of several amperes through itself, and an output stage with impedance in the thousands of ohms. The result is a voltage spike whose voltage is opposite in polarity to the previous amplifier output voltage. Diodes can snub the spike, and stave off immediate destruction of the amplifier, but not before substantial disruption to the operational environment has taken place.

When an inductive load causes an amplifier to limit current, the sudden drop in current flowing through the loudspeaker will be audible as a "click." The voltage spike (up to more than twice the peak output voltage of the amplifier) will pass through the high-pass section of a crossover network, and can overdrive or even demagnetize the upper-range loudspeakers. In addition, the rapid collapse of the magnetic field (time constant determined by the loudspeaker's inductance and resistance) can cause excess flexing of the woofer at the apex of its cone, leading to premature failure there as well.

In contrast to current-limiting, clipping due to voltage limits usually places the output stage in a low impedance state which avoids inductive "spikes." If amplifier RD overmatches loudspeaker RL, current-limiting and inductive "spikes" are avoided.

It would be helpful to have protective circuit trigger indicators on amplifier front panels, so the operator is warned of the problem before its consequences are serious. If loudspeakers and amplifiers have compatible RL and RD, respectively, then a clipping indicator is sufficient. For highest sound quality and reliability, protective circuits should not be triggered, ever. When protective circuits are not triggered, they may be harmless to listening pleasure.

RD-RL mismatch can lead to amplifier current limiting at peak outputs. When this happens, the amplifier will not be able to deliver as much undistorted power to the loudspeaker as it could to a like-impedance resistive load. The human ear has some tolerance for amplifier clipping due to either voltage-or current-limiting. If RD-RL mismatch is limited to an infrequently used band of frequencies or one that is never driven with peak power, or if the consequences of current-limiting are not too offensive to either the equipment or the listener's ear, then it may not be a severe problem. Amplifier clipping has no place in systems where the highest possible sound quality or system reliability is desired. It is probable that many cases where amplifiers "measured the same" but "sounded different" can be explained by measurable differences in RD.

Amplifier RD should be above 0.4 at the common load impedances of 4 and 8 ohms to interface acceptably with a variety of loudspeakers. It is the author's experience that a significant number of amplifiers in use fail this criteria. Loudspeaker RL should be less than 0.7 to interface acceptably with a variety of amplifiers. Published tests indicate that a significant number of loudspeakers in use fail this criteria, as well. It is probable that RL-RD mismatch is widespread. No wonder so many amplifier-loudspeaker combinations are unsuccessful!

RL and RD do not eliminate the need for earlier techniques for characterizing loudspeakers and amplifiers, and are only meaningful when used in conjunction with loudspeaker minimum impedance and amplifier output into various load impedances. Since the low-impedance cases are the worst for RD and RL, it is usually sufficient to specify amplifier RD at only the lowest impedance that it is rated to drive, and specify loudspeaker RL at the point where the quotient of RL and impedance is lowest.

Many existing systems would be enhanced by an analysis of RL and RD in conjunction with loudspeaker impedance and amplifier power output for various load impedances. Mismatch will be detected in many cases and should be followed by system modification. It is hoped that use of these parameters will lead to greater success and listening enjoyment for persons using loudspeakers and amplifiers. ■

SUSAN BOREY

Sound On Broadway

Some great things are being done with sound on the Great White Way. Find out just why actors no longer have to "shout it out."

On the surface, the responsibilities of the theater soundperson may seem minimal and undemanding. After all, theater was produced for thousands of years without the assistance of mixing boards, monitors, and microphones. Given the acoustical confines of the stage and orchestra pit, it might seem that the sound person's duty would merely consist of monitoring what the actors and musicians would yield unassisted. Actually, quantum leaps in technology and changing styles of performance have rapidly transformed the sound person, and especially the audio engineer on Broadway, into an eminent craftsman whose fine-tuned skills are indispensable to successful theater production.

Charles Bugbee III, a Broadway audio engineer whose credits include *Evita*, *Dancin'*, and *Pippin*, is part of a family whose theatrical involvement extends back to his great-grandfather, a carpenter for vaudeville shows. In his 11 years on Broadway, Bugbee has witnessed the bulk of theater sound's rapid advancement.

"The person who is really responsible for bringing high quality to Broadway is Abe Jacobs. He brought studio and rock 'n' roll techniques into theatre," Bugbee said. "His background was in rock 'n' roll, working with Jimi Hendrix and the Mamas and the Papas, but I believe the first show he came in on was *Hair*. Prior to that, sound usually consisted of running cued tapes, setting up a few foot mikes, and leaving them alone. When I did my first show, *Pippin*, I had old Altec mixing boards with 5 inputs/1 output. I usually used three of them strung together."

Today, it's not uncommon to find 50 channels in use for a show, usually through customized boards. The influx of new hardware into the theater doesn't reflect the whole scope of changes in theater sound, and arrangers, and the changing tastes of the audience.

"Broadway is no longer presenting Ethel Merman-type singers who don't need mics. The public just doesn't want it," claimed Bugbee, alluding to the new demands that singers, still mobile but often unable to project enough, unamplified, to cover a large theatre, have made on technology.

Today, many arrangers have built-in expectations for both equipment and audio engineers.

"It's not uncommon for Broadway arrangers to also work in recording studios or to arrange for television and film," said Bugbee. "Arrangers will stand by me when I'm working out a new show and say, 'Catch this line I've got going on the acoustic guitar, it's really hip. Bring it up a little bit.' The minute you've got an acoustic guitar that has to be featured over a brass section, you need a sound system that allows you to mix like that, and the sound person then becomes a creative force in the show."

UNSUNG HEROES

A prevailing irony with most sound people is that they are least noticed when performing at their best. Certainly true with Broadway audio engineers—their task is further complicated by the need for transparency of the sound system as well. The essence of theater sound is that system and technicians are providing reinforcement; people in the audience should perceive that they are hearing the actors from the stage, not from the speakers. This shift in aural perspective is one of the main differences between theater sound and any other type of audio. Broadway plays, housed for many years in intimate venues in mid-town Manhattan's theater district, are now spilling into larger facilities which often have a doubled audience capacity. Mixed emotions exist regarding the exodus into the modern, more spacious venues; the ability to present Broadway to more people weighs against a loss of intimacy and the unique charm afforded by the smaller houses which can seat a smaller audience closer to the stage. One thing is certain—the larger houses don't facilitate the sound person's attempt to sustain an aural illusion.

"I prefer the smaller houses," declared Bugbee. "Once you get into a place that seats 2,500 people, it's hard to pretend that the sound isn't coming from speakers."

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The goal of the Broadway sound person is to support the motion onstage, and this is accomplished through a spectrum of approaches that ranges between subliminal and blatant. If the intensity of a scene is building, a sensitive sound person can help the momentum by bringing out the underscore, easing it up over several minutes, if necessary. Sometimes exaggeration is the best insurance for getting the message across.

"You are mixing for people who are going to hear something once," Bugbee explained. "You've got one chance to get through to them. In a number where dancers are flicking their wrists, I would mix a woodblock up to accent the choreography. This is something you'd never do on a record; a loud woodblock would be too much. But it makes sense when it's illustrating what 20 dancers are doing. A lot of little accents like that are written into the score to support what's going on onstage."

THE SOUND PERSON AS ACTOR

The full comprehension of a Broadway show that permits broad reinforcement and concise accentuation requires a massive commitment from the sound person. Basically, the entire show must be memorized. This includes all of the actors' lines, the way each of them is delivered, and the position of delivery onstage. The musical score must be committed to memory as well, including individual instrumental parts as well as their interrelations in the orchestration.

"You have to totally immerse yourself in it. You have to be totally alert all the time," Bugbee declared. "You can develop amazing powers of concentration. It probably takes six to eight weeks to do a show well, to get the most out of the orchestration, the lines, the EQ, the echo, and everything else. When I start a new show I always get a copy of the score, and that's what I mix to until I know it by heart."

Despite the lengths the sound person goes to in preparation for an ultimately sensitive presentation, there's no guarantee that any show will run long enough to fulfill the massive investment of time made.

Although the sound person is responsible for making the show sound the same, in its essence, night after night, the consistency is dependent on flexibility.

"When I'm teaching people a show, the minute I see them writing down numbers off the board I know they're going to be in trouble," said Bugbee. "Numbers don't mean anything from one night to the next. They can serve as ball park references, but you've got to constantly listen to everything and make adjustments. If the dancers look a little sloppy, maybe they're not getting enough snare drum in the monitor and so they can't hear the beat well enough. If a singer's intonation appears to be a bit off, perhaps they need more piano and less drums in their monitor. If a trumpet player had a fight with his wife the previous night and he's playing four times as loud as he ever has, you've got to completely disregard what number the fader is on. Sometimes actors try out their lines different ways and you've got to be right there to compensate, if necessary. Sometimes substitutes are in the orchestra, and they play a part very differently."

Even the adjustment of equalization, often a once-a-night chore for a sound person, does not remain constant for the Broadway sound engineer.

"We make hundreds of EQ changes during a show to compensate for things like wireless mics being used in different costumes, or when a flute comes up to fill a two-bar hole and then must instantly fit back into the orchestra. When someone delivers a line from a balcony, the footlight mics are EQ'ed differently than when an ensemble is singing into them at the front of the stage.

The audience provides yet another reach of opportunities for flexibility. There's an interplay with the audience that a good engineer should feel and use to make fine adjustments.

"You can tell when you've got an audience going with a big musical number," said Bugbee. "Sometimes you've got to wake them up a bit, so you punch the trumpets a little more than usual. You also have to have a sense of climax, a feel for the crescendos in applause, and know how to mix around that."

ONE MAN SHOW

Usually, one person is in charge of all the audio for a show. This includes mixing the stage, the orchestra, monitors, several sets of main speakers which often include delayed speakers in the mezzanine and/or balcony, dressing room speakers, and running tapes which often accompany parts of a show. The sound person also works very closely with the director, composer, and arranger to bring out the salient points and subtle nuances they have in mind.

A Broadway audio engineer strives for naturalness and transparency working within a dynamic spectrum that ranges from having the system completely off to making musical numbers sound as big as possible. Broadway sound systems are thus designed with attention to their unique requirements for performance. Usually, masked towers of speakers flank the stage for the main punch of musical numbers. Front fill speakers, set to benefit audience members seated front center, out of the focus of the main speakers, will also help fill out the corners of the theater. If the orchestra is located in a pit at the edge of the stage, front fills will boost the sound from the stage over the instruments. The delayed mix for under- and over-balcony speakers is done separately; instruments carry naturally in the theater to different distances, and each speaker mix necessitates a different perspective of aural priorities.

Monitors, crucial for precise synchronization between actors and orchestra, are always out of view of the audience. Usually columns rather than wedges, most often there is a pair upstage and a pair downstage.

THE MIC PROBLEM

Although microphones are a relatively new tool in the theater, the flexibility and staging possibilities they've afforded have already made them indispensable on Broadway. Onstage, vocals are reproduced by the wirelesses and unidirectional mics (dubbed "footlights" by their position at the edge of the stage). Commonly Sennheiser 816 shotgun mics or AKG 451s, footlight mics are subject to very exacting equalization because feedback can be a big problem. Another common problem with floor-mounted microphones comes from the excessive noise produced by the footfalls of actors and dancers. The situation can be remedied, to a certain extent, by a combination of careful mounting and sensitive equalization. Overhead and movable shotgun microphones don't work very well onstage; they pick up the monitors and can deliver a very muddy sound.

Wireless microphones, permitting ultimate mobility to actors, and allowing them to whisper their lines, if desired, are currently less reliable than might be hoped for. The greater the number of wireless mics running, the more they will interfere with each other. The mics are small, but must be attached to battery packs. They don't always afford the best position for mic'ing a singer since they must be concealed in the costume.

"One problem with wireless is that if a performer sweats a lot, the mics can short out. This usually happens with dancers," explained Bugbee. "One safeguard in this situation would be to mount two wireless mics in that performer's costume. This necessitates very quick interchanging between costumes. Frequent handling of these mics increases the chance of malfunction. Usually it's not the transmitter that breaks, but rather the microphone or antenna that dangles from the battery pack. I generally have someone running around backstage with spare mics, antennas, and battery packs, and ideally, by frequent, rapid testing you'll find the problem before the audience hears it. Everything breaks eventually, but you just hope it goes when you're checking it."

Diligent as they may be, the sound people don't always catch problems in time. However, there are procedures that, through quick thinking and creativity, can help salvage a potentially disastrous moment.

"The performer usually can tell that their mic isn't working by the way it sounds through the monitors," Bugbee said. "Often they can move up to the footlight mics, and sometimes they can move close enough to another performer's wireless to be picked up on that."

Tape machines are often used on Broadway to provide sound effects and additional music. Cart machines are usually employed; although the sound quality may not be as high as reel-to-reel, cartridges are much easier to handle. The sound person's time is at a premium when mixing a show, and it's necessary to be able to pull a cart out, pick up the next one, and know that it's ready to go without cueing. Double tapes are usually run on two decks with identical tapes; if a tape breaks or transformer goes, there's a good chance the sound person will be covered.

Stereo, a development that has served to enhance recorded music, doesn't readily benefit theater sound, except when used for an occasional effect. The primary drawback of stereo on Broadway is that there are never enough people sitting in the center to benefit from it. If an actor is moving across the front of the stage, one might assume that the effect would be enhanced by panning the footlight microphones in the direction he is walking. However, the audience on the side of the stage he starts from doesn't need more reinforcement to hear him; the other side does. In order to insure complete coverage with a stereo system, you must actually pan the mics in the reverse direction.

Despite the dedication, extreme precision, and creatively flexible technical acumen required from the Broadway audio engineer, their contribution to theater arts is generally unlauded. In the eyes of the union, the sound person is still legally considered an assistant electrician. No Tony award is given for excellence in this facet of theater, even though the Broadway audio engineer, who mixes live the equivalent of a double album each performance, is as indispensable as any key figure in a production. An excellent sound job can't make a poor show sound good, but poor sound can certainly ruin a good one.

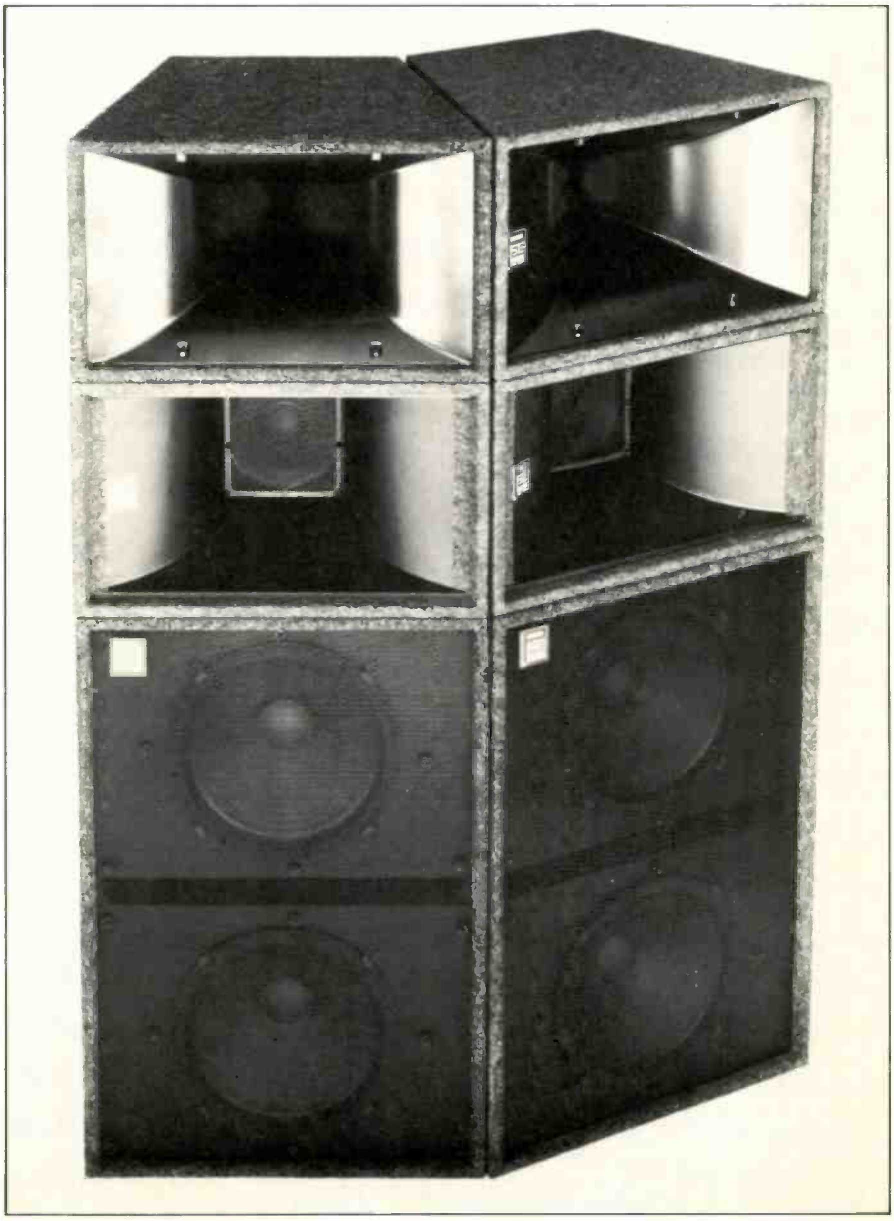
In the next issue, db visits Broadway to report on the sound systems and special audio engineering techniques employed with several Broadway shows. ■

COMPUTER-DESIGNED COAXIAL

• Cetec Gauss' Model 3588 is said to be the first computer-designed coaxial loudspeaker. It features a convervative power rating of 200 watts RMS, and has been tested to deliver clean sound at continuous program levels of 750 watts. Metric sensitivity is 95 dB for the low frequency and 109 dB for the high frequency. Because of proprietary design parameters, both drivers are virtually in the same acoustic plane, satisfying the Blauert and Laws criteria, thus eliminating the need for time compensation networks. The new coaxial is designed to work with any standard professional quality crossover. A new Computer-Aided Time-Spectrometry program was developed to design the unique cosh horn for the coaxial. This design provides an extremely stable image, reduced second-harmonic distortion, and virtually no midrange shadowing.
Mfr: Cetec Gauss

Circle 50 on Reader Service Card

FULL RANGE, TIME-CORRECTED PA SYSTEM

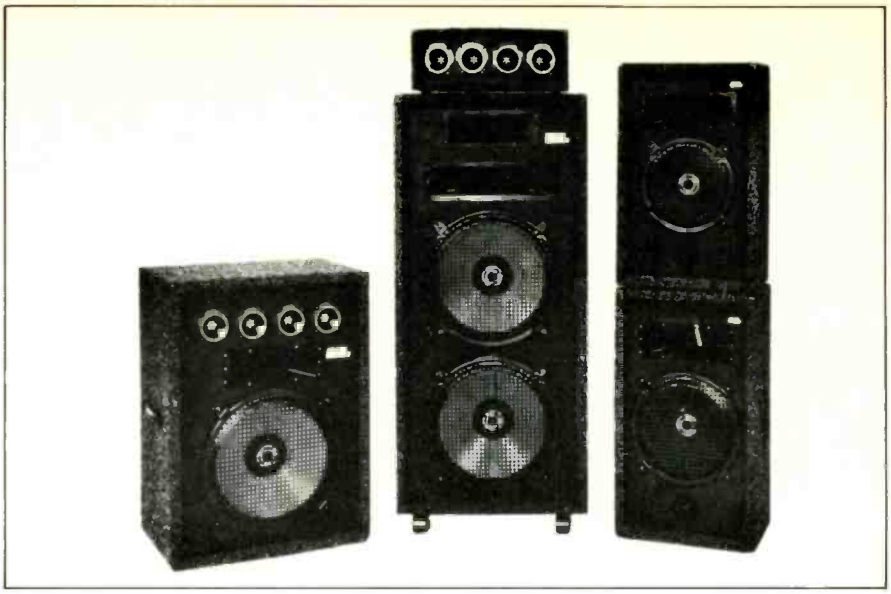


• Professional Audio Systems' MRS-1 (Modular Reinforcement System 1) is a three-way, modular, full range, time-corrected PA system comprised of a double 18-inch bass module, a single 12-inch mid-bass module, a constant coverage horn with a 2-inch compression driver, and dedicated electronics. Controlling the system is the one-rack space System Processor. This unit is internally calibrated for the Time Offset Correction™, the crossover filters, and the turn-on turn-off transient protection. Other than the On/Off switch, output levels, and limiter controls, everything is permanently set. Cabinets are constructed of ¾-inch multi-ply with 2- × 3-inch bracing and are wedge-cut for clustering. High-density weave exterior adds to system durability. Frequency Response is 40 Hz to 15 kHz, ±3 dB; Phase Response is 100 Hz to 10 kHz, ±10 degrees.
Mfr: Professional Audio Systems
Price: MRS-1 with mono System Processor, $4,160;
MRS-1 with stereo System Processor, $4,660.

Circle 31 on Reader Service Card

NEW ROAD SYSTEMS

• American Acoustic Labs' new Road Systems consist of a five-model loudspeaker line for professional applications. The new Road Systems line, comprised of models RS-115, RS-215, RS-TA4, RS-112, and RS-110, offers four premium sound reinforcement loudspeaker systems and a dynamic tweeter array for a variety of professional applications. Each new AAL Road System model is housed in a carpeted enclosure designed to enhance acoustic performance. Features include quality components such as polypropylene and paper drivers engineered by pro driver manufacturing affiliate MTX. The RS-215 also uses MTX's CD-60W-8 compression driver in conjunction with a 4- × 14-inch pro lens for extended frequency response and low distortion. For protection against transit damage, MTX drivers used in the AAL Road Systems feature MTX's metal-mesh grills. The drivers and grills are also fastened by four "quick clips" for fast maintenance. Other features include constant directivity horns made from non-resonant polyurethane for cleaner sound reproduction with less equalization. Each Road System model has thermal-protected high-frequency and midrange components for long-lasting performance. Power handling capability of the new AAL Road Systems ranges from 100 to 200 watts-per-channel RMS.



Mfr: American Acoustics Labs (AAL)
Price: From $249 to $699

Circle 32 on Reader Service Card

REFERENCE MONITOR SYSTEMS

• Fostex has incorporated its Regulated Phase Technology in the tweeters of the new RM 765 and RM 780 monitors. The essential idea of this RP transducer is a flat, thin film diaphragm onto which the voice coil is printed. This assembly is then suspended between magnetic circuits arranged on both sides of the diaphragm, with the same polarities facing each other and the opposite polarities adjacent to one another. The end result of this arrangement permits the diaphragm to be driven with absolute uniformity; it provides the same uniform frequency response across an extremely wide dynamic range. The double spider design of the RM-Series woofers is the perfect complement to the RP tweeter. In a conventional single spider design, the compliance from one direction differs from the other. This difference produces unacceptable distortion, particularly in the second-harmonic region. Fostex engineers added a second spider in a push-pull complementary configuration to cancel the compliance effects of a single spider design. The result is a 10 dB improvement in second-harmonic distortion. A two-inch voice coil/magnet assembly and a computer modeled bass reflex design complete the profile of what is definitive low-end response in the Reference Monitor Series. These two drivers are first placed in a coaxial relationship, then adjusted for time compensation in a true concentric design. Regardless of placement, the Fostex 765s and 780s may be switched (via the front panel) for proper response. Since they are time and phase coherent, they can be mounted vertically or horizontally. Rack adapters and mounting hardware are available, and all holes have been pre-drilled for convenient interface with any standard rack.

Mfr: Fostex Corporation of America

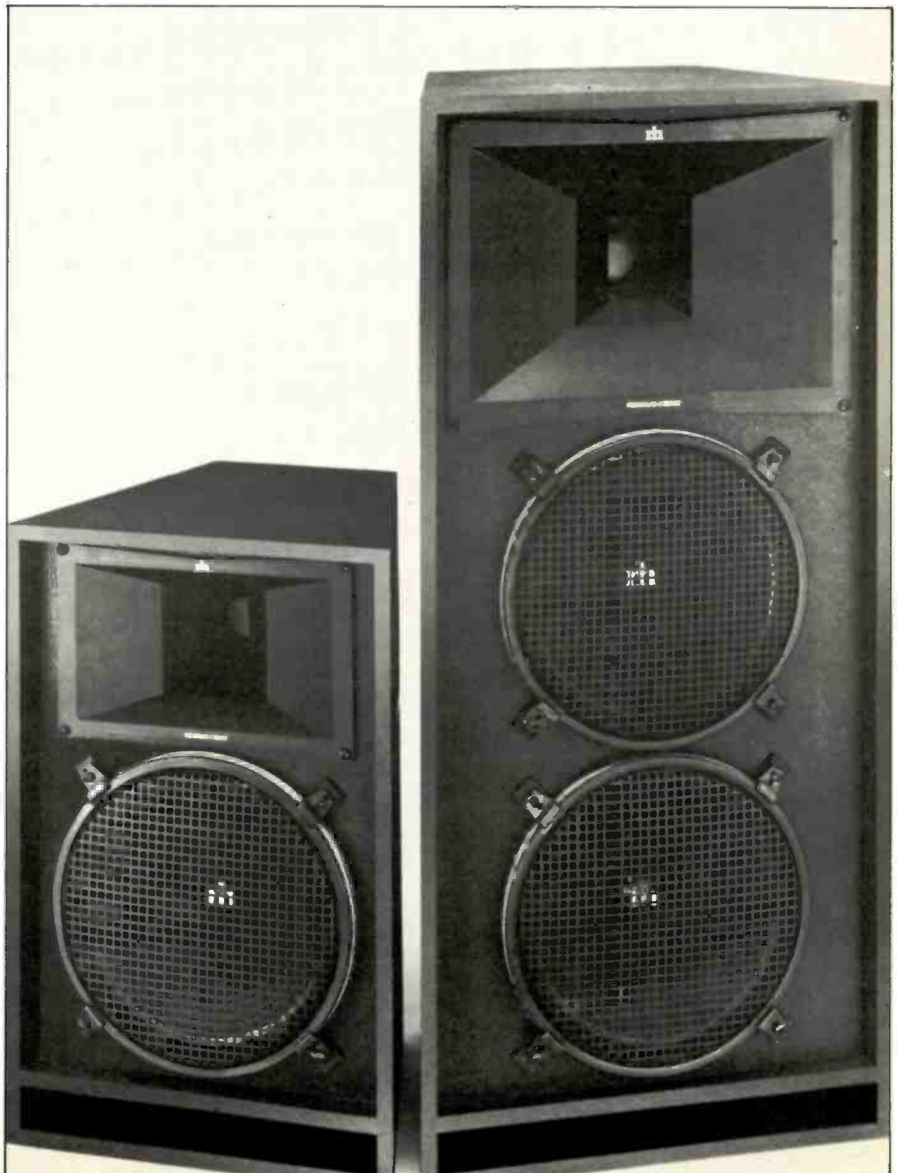
Circle 33 on Reader Service Card

DIRECT REPLACEMENT LOUDSPEAKERS

• MTX's new PL-5 and PL-5G loudspeakers are designed for powerful output, efficient response, and wide dispersion. The PL-5 is the newest addition to MTX's Pro Line Series of technically advanced drivers for professional and commercial applications. The new driver is also offered as the PL-5G model, which includes an integrated metal grill, precision stamped for maximum protection. Ideally suited as a direct replacement driver for in-line arrays and ceiling installations, the PL-5 and PL-5G can also be used as midrange drivers in monitor systems. Utilizing a rugged die-cast basket design, features include a 4½-inch black paper cone with treated cloth and plastiseal surround. The midrange is driven by a one-inch ferrofluid, Kapton voice coil with a 12-ounce magnet. Developed for wide dynamic range, power handling capacity is 100 watts-per-channel RMS. Frequency response is 164 Hz to 13.5 kHz. Sensitivity is 95 dB SPL at one meter with a one-watt input. With an 8-ohm impedance, the resonant frequency is 164 kHz.
Mfr: MTX
Price: PL-5, $49.00 suggested retail; PL-5G, $59.00 suggested retail

Circle 34 on Reader Service Card

MOVIE THEATER SYSTEMS



• Renkus Heinz' two new speaker systems, Model SCS 2552-CB and Model SCS 1582, are designed for today's movie theaters. Both models are two-way designs, utilizing components with proven reliability. The units have very even frequency response, high efficiency, and wide dynamic range. R-H constant directivity horns provide ideal audience coverage. The compact enclosures are designed for behind-the-screen or surround installations. Model SCS 2552-CB utilizes the 200-watt SSD 3301 2-inch compression driver on a CBH 500 (constant directivity) horn and two 15-inch low frequency drivers. This system can generate 132 dB SPL from 40 Hz to 17 kHz. The smaller model SCS 1582 is similar in performance, but uses one 15-inch bass driver and has a lower frequency limit of 45 Hz. Both models require substantially less equalization and provide more uniform, controlled coverage than traditional cinema speaker designs. Additionally, both models are compatible with the unique R-H "smart" crossovers.
Mfr: Renkus Heinz

Circle 35 on Reader Service Card

THREE-WAY "SYNTHABLE" SPEAKER SYSTEM

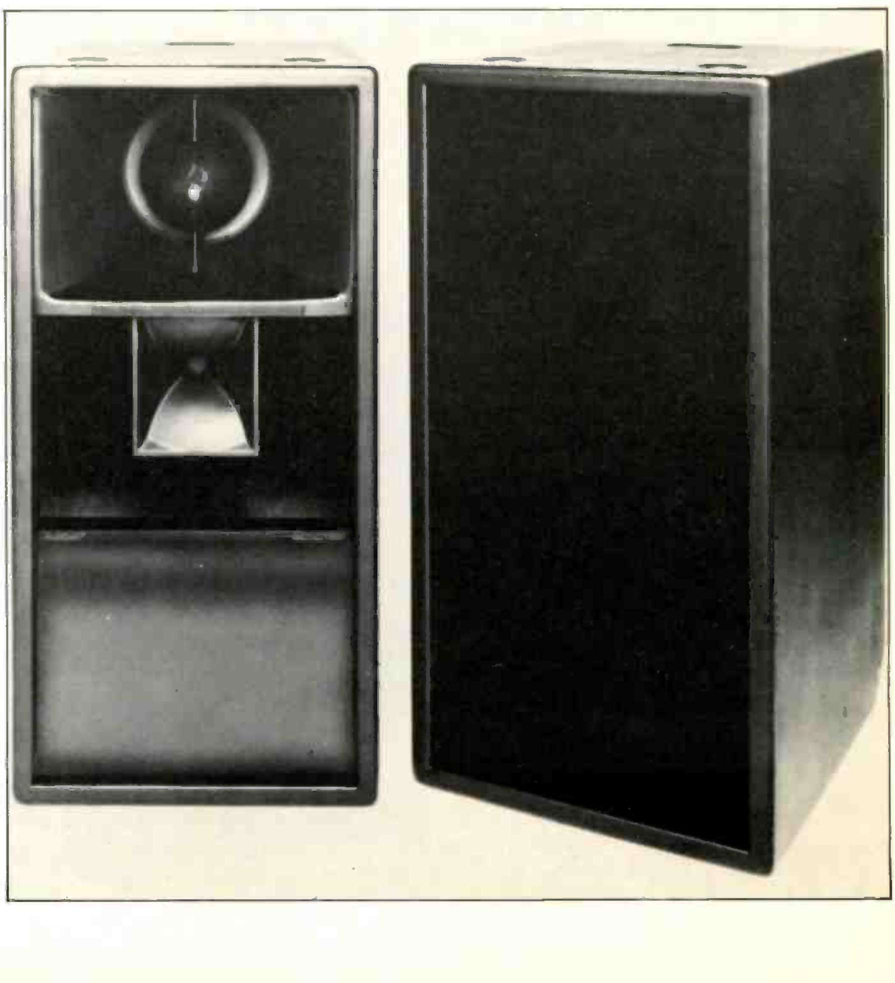
• TOA's 380SE is the first product in a line of gear designed for professional applications that involve electronically-created music and sound. The 380SE is a powerful three-way system that adds no coloration to the original audio. A smooth, extended frequency-response and controlled directivity are benefits of the 380SE configuration, which incorporates an exponential horn tweeter and a Theile-Small aligned bass reflex design (providing greater efficiency and a wider bass range). The 380SE is constructed entirely from TOA's highest quality components. It provides continuous high power handling of 360 watts. It contains full range inputs, bi-amp and tri-amp connectors, and four bridging connectors. Flush-mounted on the upper side of the speaker are two level controls, one for mid and one for high frequencies. This allows tailoring of the 380SE output to performance requirements and room acoutics. TOA's new speaker reproduces all types of sounds at all levels with sonic accuracy. The 380SE illuminates subtle variations in pitch and level, whether handling a single note or a full synthesized chorus.
Mfr: TOA Electronics

Circle 36 on Reader Service Card

THREE-WAY LINE ARRAY SOUND REINFORCEMENT MODULE

• Turbosound's new TMS-2 is the latest in a series of three-way professional sound reinforcement speaker enclosures. Designed for bands, clubs, discos, concert halls, or any other application requiring exceptional fidelity, ease of handling, and compact size, the TMS-2 is highly efficient with 103 dB (1 watt, 1 meter) average sensitivity rating and a maximum SPL of 131 dB (peak). It will reproduce low frequency information to 60 Hz (-3 dB) (preliminary spec), yet the total enclosure measures only 34 × 17 × 23 inches. The unique design of the TurboBass™ device makes this combination of power and compact size possible. Like the other enclosures in the TMS series, the TMS-2 uses the patented TurboBass™ and TurboMid™ devices to cover the low and middle parts of the sound spectrum. Highs above 4 kHz are handled by a one-inch compression driver and exponential horn. The patented TurboMid™ device uses a horn-loading technique that enables the proprietary Turbosound 10-inch driver to cover the full midrange spectrum from 250 Hz to 4 kHz. Unlike the typical pro audio system in which two drivers of differing materials are responsible for the vocal range, the Turbosound design eliminates the midrange crossover point and its attendant problems such as phase cancellation, coloration, and uneven frequency response in the most prominent part of the sound spectrum. The TurboMid™ device also enhances the sound of the high frequency driver by taking over the most difficult of its operating range, resulting in a smoother, more natural-sounding top-end and a more reliable compression driver. The three components of the TMS-2 are passively crossed-over and arranged in a phase and amplitude-aligned design that reproduces transient peaks with extreme accuracy. This enables the TMS-2 to deliver clean, "alive" sound that transmits the energy of a live performance directly to the listener without the filtering and blunting of impact typical of the usual PA sound.
Mfr: Turbosound
Price: TMS-2, $1,530 each

Circle 37 on Reader Service Card



ARTICULATED ARRAY® LOUDSPEAKERS

• The Bose 402 and 402-W are actively equalized loudspeaker systems designed for high-quality reinforcement of voice and music. The 402 speaker is ideal for applications requiring a rugged, portable enclosure, while the 402-W speaker is intended for use in permanent indoor sound system installations. The acoustic properties of the 402 and 402-W systems are identical. Both speakers employ four 4½-inch Bose D-22A high-sensitivity drivers, mounted vertically on a faceted Articulated Array baffle assembly. The drivers feature low-impedance edgewound aluminum voice coils, 12-ounce Ferrite V ceramic magnets, molded polyester frames, and an advanced cone and motor system for high linear excursion capability and power output. Tuned Reactive Radiator slots reduce distortion by controlling the cone excursion required to reproduce mid-bass frequencies. An Acoustic Diffractor broadens and smooths the horizontal radiation pattern of the inner drivers for more uniform side-to-side room coverage. A built-in Directivity Control circuit maintains the vertical dispersion pattern through the high-frequency range and also protects the drivers from the effects of high-frequency overload. The 402 speaker enclosure is composed of polyethylene copolymer structural foam, reinforced with 10 percent mica for improved durability and impact strength. The weather-resistant design of the D-22A drivers allows the 402 system to be permanently mounted outdoors in a wide variety of climates without damage. The Reactive Radiator slots provide effective enclosure drainage. The acrylic-coated walnut-grain finish on the 402-W enclosure can be painted to match special color requirements. The complete 402-W baffle assembly can be easily removed from the wood cabinet to facilitate the installation of mounting hardware. The fixed 402-E Active Equalizer assures smooth, accurate spectral response across the entire operating range of the system. Sharp subsonic and ultrasonic band-limiting filters reduce power waste, stage noise, high-frequency instability, and interference. Two independent signal channels are provided in a compact unit that fits into two spaces of a standard 19-inch equipment rack with the optional RMK-4 Rack Mount Kit.

Mfr: Bose

Circle 38 on Reader Service Card

HIGH FREQUENCY COMPRESSION DRIVERS

• The new Emilar professional series EC 320A and EC 314A high frequency compression drivers incorporate the latest as well as proven materials, technology, and manufacturing techniques. Each diaphragm is individually spun from structural aluminum, and then, for added strength, is heat-treated. Structural aluminum in this application produces optimum frequency response, high durability, and sonic excellence. The non-resonant polyimide suspension provides structural and physical stability and superior damping properties with high fatigue strength. The phasing plug, a classic concentric slot design, is cast from non-resonant, non-metallic material. Precise assembly procedures and individual testing guarantee that each driver meets demanding quantitative and qualitative standards.

Mfr: Emilar

Circle 39 on Reader Service Card

LIVE PERFORMANCE LOUDSPEAKERS

• Tannoy's Wildcats are a new range of live performance loudspeaker systems. Their modular construction ensures that whether you use two or 22 enclosures, the system adapts itself to the artists' needs. Where the venue or program material calls for low bass response, the Leopard bass system can be used, and unlike most so-called "modular systems," does not necessarily need bi-amplification as it incorporates its own passive crossover. This enables one amplifier to drive, for example, two Leopards and a Lynx. The Wildcats are a unique development aimed at three significant market segments, the first being the area of group PA where they would be a natural choice for cabaret artists who require clarity and flexibility from their system, as well as ease of installation. The systems are also suitable for larger venues where the artists' music requires true low bass response without the irregular response normally associated with horn-loaded bass systems. The second market segment is the discotheque and club industry. The Wildcats can be tailored to suit most environments and provide high levels of sound with characteristics normally associated with studio monitoring. The third market is Audio-Visual systems. Whether it be a small portable system using the Lynx or a major project using multiples of enclosures, the Wildcats can provide a consistent character of sound.
Mfr: Tannoy-Crown

Circle 40 on Reader Service Card

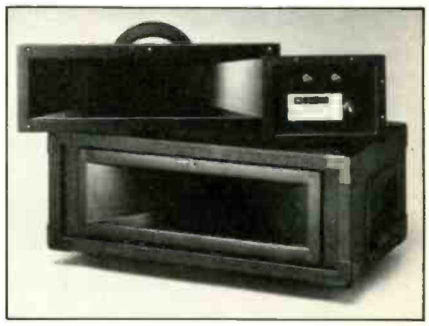
PROFESSIONAL MONITORS

• Carvin's new 750-M Monitor System delivers excellent tone quality that can be easily heard by the performer. A deep bass response and 125 WRMS power-handling capacity is delivered by the Magna-Lab 12-inch 1224 speaker. The high-end is handled by the new Magna-Lab 490 horn driver giving crisp, clear reproduction. A frequency response out to 20 kHz maintains harmonic accuracy so that there is a clear definition of the sound. The featured XC1200 crossover is a full 12 dB/octave frequency divider that assures a smooth, accurate transition from the woofer to the horn. The optional Electro-Voice EVM-12L speaker may be ordered in the 750E models, affording slightly more handling capacity.
Mfr: Carvin
Price: $169.00 pro net

Circle 41 on Reader Service Card

HIGH FREQUENCY SYSTEM

• JBL Incorporated's MI Series Model MI-261 High Frequency System and Model MI-291 High Frequency Power Pack are the result of an increased demand for separate availability of the flat-front Bi-Radial™ horn, the titanium diaphragm compression driver, and a network with high-frequency equalization and level switching. While the MI-261 packages the components in a portable enclosure, the MI-291 has separate components for custom enclosure mounting. The flat-front Bi-Radial horn is a patented JBL design which eliminates the problems of midrange narrowing and high-frequency beaming associated with conventional horn designs. It has 90 degree horizontal by 40 degree vertical nominal coverage with uniform on- and off-axis frequency response in the horizontal plane from 630 Hz to beyond 16 kHz. The high-frequency driver features a pure titanium diaphragm with JBL's patented diamond-pattern surround. The high stress limit of titanium, together with the surround, allows the design to achieve the performance of more expensive and fragile exotic materials while maintaining high reliability and ruggedness. The driver uses a high-energy strontium ferrite magnet for maximum efficiency and extended response. The network is tailored to optimize the crossover response and uses quality electronic components, including non-inductive, non-polarized mylar capacitors and air-core inductors. A three-position level switch allows level matching to low-frequency systems of varying sensitivities, and high-frequency equalization provides maximum response flatness at each level setting.



Mfr.: JBL Incorporated.

Circle 42 on Reader Service Card

COMPUTERIZED AUDIO INFORMATION RETRIEVAL SYSTEM

• Gotham Audio Corporation's SYSTEX™ is a fully computer-based audio system for broadcast, TV, film, and recording studio use. The SYSTEX (a 330-Megabyte hard disk storage system) digitally records, plays back, and locates audio information with accuracy and speed. The system was created specifically for the needs of radio broadcasters and film/video editors who require instant random access to a large library of audio material for news broadcasts, commercial spots, dialogue, music, and sound effects. For broadcasters, SYSTEX represents a new era in station production and information storage, and could ultimately replace the standard cartridge machine because of its maintenance-free operation, high reliability, and digital audio quality. And, since the information is stored in a central and accessible location, all broadcasts can also be tracked, logged, and monitored for accounting or scheduling purposes. Recording engineers can utilize SYSTEX as a replacement for cumbersome sound effects libraries and as a controllable command center from which one can assemble and edit soundtracks. Access between selections on a disk is made without audible interruption of the audio signal.

Mfr: Gotham Audio Corporation
Price: Basic dual-rack system (including CPU, single hard disk, and sequencer), $125,000
Additional sequencers, $35,000 each
Additional Winchester disk drives, $10,000 each

Circle 43 on Reader Service Card

MEASURING COMPUTER

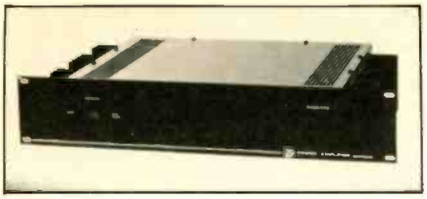
• Ortofon has introduced the second generation of its P400 Measuring Computer. The current device incorporates a number of extensions and improvements, making the computer even more qualified for routine testing of audio transducers and systems. The P400 Measuring Computer offers automatic in-line testing of performance data of loudspeakers, headphones, microphones, and dynamic transducers used in telephone equipment. The measuring parameters are frequency response, sensitivity/efficiency, rub & buzz, polarity/electrical phase, and impedance. The minimum test cycle for a complete test program is four seconds. The instrument is supplied with the necessary software that allows the user to establish his own references and tolerance bands from the front panel keyboard. All stages of the test program are stored on an exchangeable EE-PROM cassette. In the automatic operation mode, the front panel keyboard is not operational; the instrument is controlled directly from the EE-PROM cassette. If changes are required in the program, a trained operator can reactivate the front panel keyboard, display the program details that require changing, and make the necessary corrections. Any

establishment or modification of test programs is easily carried out. Possible programming errors are displayed on the screen. The test results are displayed as a GO/NO GO signal or as curves with their associated reference or tolerance band. The test results and all individual data can be printed out by a thermal printer/plotter and/or transferred via an IEEE 488 interface into an external memory for subsequent statistical evaluation. An external memory, such as a floppy disk, can also contain test programs as well as additional software for other manipulations and calculations. This allows true production management procedure, as variations in performance data instantly show up as deviations from references and established tolerances. Immediate action can then be taken to avoid further deviations from established production standards. Orotofon's new P400 Measuring Computer replaces the traditional listening test of audio transducers with an objective measurement of relevant performance data.

Mfr: Ortofon Instruments A/S

Circle 44 on Reader Service Card

NEW POWER AMPLIFIERS

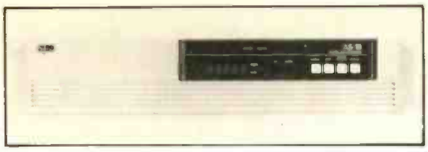


• Protech Audio Corporation has added two new power amplifiers to their line of professional sound equipment. Designated the Model 874 and Model 875, the new units are capable of delivering 60 watts and 125 watts, respectively. Both models are designed to provide top quality professional specifications and reliability at highly competitive prices. Each of the two units provides an 8-ohm, 70.7-volt transformer-isolated output. The units may be ordered with either a 600-ohm or a 10-kilohm transformer-isolated, balanced input. Power to the units may be supplied by either 117 VAC or 24 VDC. The use of a toroidial power transformer greatly reduces stray magnetic fields often generated by power amplifiers, and allows the units to be mounted in closer proximity to low-level equipment. The package size for both units is 19 inches wide by 3.5 inches high by 10½ inches deep. Fusing of the AC line is accomplished via a front-panel-mounted On-Off switch/Circuit Breaker. Additional features include: Frequency Response of 30 Hz to 20 kHz, ±1 dB, Distortion of 1.5 percent maximum at full power, and a front-panel-mounted Overload Indicator.

Mfr: Protech Audio Corporation

Circle 45 on Reader Service Card

DIGITAL AUDIO SYNCHRONIZER



• NEC's new AS-18 Digital Audio Synchronizer can be used in any audio delay or audio timing application. In its standard configuration, the AS-18 can delay the audio signal up to one second. With optional memory, it can delay up to four seconds. When used with the NEC FS-18 Frame Synchronizer, the AS-18 can provide automatic compensation for audio-to-video delay. Otherwise it can be combined with any other frame synchronizer on a manual basis. Features include: two audio channels—third channel capability available with optional Audio-2 card; audio delay time up to a maximum of one second per channel (standard); delay step—1 millisecond (with optional memory, delay can be increased to a maximum of four seconds per channel); the AS-18 can be used with any frame synchronizer, but when combined with the FS-18, no manual adjustment is required; up to three audio channels can be controlled independently with optional controls (optional remote control panel is available); and in non-video applications such as radio broadcast programming, the AS-18 can be used as an audio delay line with a delay capability of up to a maximum of 12 seconds. This can be useful for monitoring and modifying (spot erasing) live radio broadcasts. The AS-18 features 16-bit quantization and a sampling frequency of 48 kHz. Frequency response is +0.5, -1.0 dB, 20 Hz to 20 kHz; the dynamic range is better than 90 dB. Audio output level is -20 dBm/0 dBm, 600 ohms balanced. THD is below 0.05 percent.

Mfr: NEC America, Inc.

Circle 46 on Reader Service Card

AUDIO TIME COMPRESSOR/EXPANDER

• Lexicon's new Model 1200C Audio Time Compressor communicates with a wide variety of one-inch VTRs and editors via an RS-422/232 port. This communications capability allows the 1200C to tie into station automation and to respond to remote instructions. The 1200C can communicate via the Sony BVH-2000 protocol to VTRs and tape editing systems that offer time compression/expansion editing software. The Ampex VTR interface greatly improves servo lock-up time, reducing it to less than three seconds. As a result, the pre-roll requirement is dramatically reduced. The 1200C's timing capability has been improved to better than one second per hour of play time. Improved input level-matching and signal-to-noise characteristics are optimized for broadcast applications. An optional RS-422/232 serial data communications board is required for the Sony BVH-2000 protocol feature. A field upgrade for earlier 1200 versions is also available.

Mfr: Lexicon
Price: 1200C, $8,500
Communications board, $1,000

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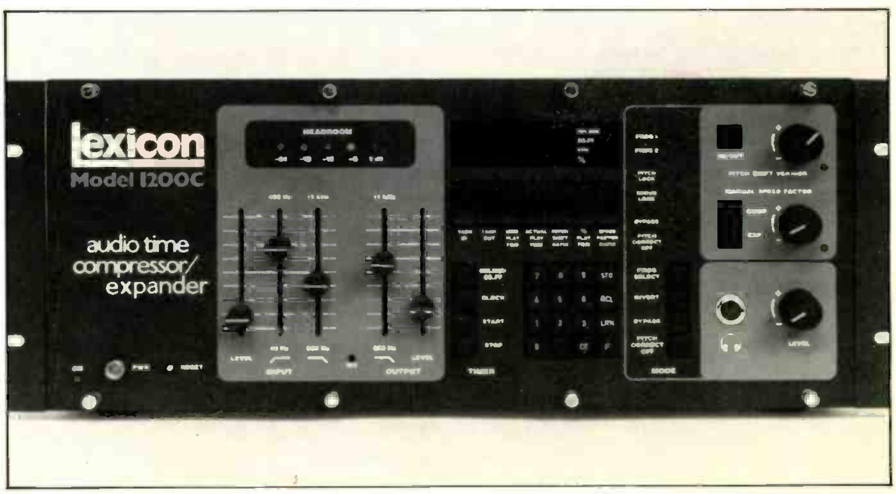

• **George Massenburg Labs** has completed installation of the first of three new moving-fader automation systems at Conway Recorders in Hollywood. Two other systems are slated to be installed at leading audio and film studios within the next six weeks. The custom-built Conway system includes automated operation of 40 input faders, eight echo returns, six groups and a stereo fader, all with automated muting. GML engineers installed the system on a Neve 8108 console.

• **Glenn H. Derringer** has been named vice president for sales and marketing at **Kurzweil Music Systems**. Derringer previously served in various marketing capacities with the Wurlitzer company and Baldwin Piano and Organ company over a period of 15 years.

• Ioan Allen, vice president of marketing for **Dolby Laboratories Inc.**, has announced a new position and increased responsibilities for **Scott Schuman**. Schuman's new title is director of market development. His responsibilities will include marketing and promotional support for licensing and engineering for Dolby Laboratories. Some of the areas he is presently involved with are Dolby HX Pro "headroom extension," Dolby C-type software, Dolby noise reduction for VHS precorded videotapes and the Dolby Digital Audio System. An expanded program to extend communications with the US-based marketing arms of overseas licensees will come under his direction. Schuman will continue to be located at Dolby Laboratories' San Francisco headquarters. Schuman will also be involved with long term market planning and development support for a variety of concerns at Dolby Laboratories, including market research in both the consumer and professional areas.

• **Audiotechniques, Inc.** executive vice president, **Robert Berliner**, has announced the appointment of **Geoff Hillier** as technical services manager. Hillier, who was formerly director of engineering for Trident, USA, will have management responsibility for installation, service and equipment maintenance and parts for Audiotechniques, Inc.

• Since opening two and a half years ago, **Triad Studios** in Redmond, Washington, has built a reputation as one of the premier recording studios of the Northwest. Recently, due to the heavy demand placed on Studio A and the need to provide clients with an increasing range of equipment and service, Triad has begun construction of Studio B. In the new Studio B complex, the lounge, maintenance area, and storage room are complete. The "floating" concrete floor is in place and the wall and ceiling construction is now underway. Studio B will incorporate the same high quality acoustical treatment and attention to detail as in Studio A. In addition there will be the same feeling of spaciousness, with a total of 5,000 square feet of studio, control room and lounge area available. The 24-track rate for Studio B will be competitive with many 16-track facilities and, in addition, Triad will have 160-track recording and a comparable rate available.

• Noted recording specialist **Michael Szakmeister** of Brookline, Massachusetts, has been appointed to the faculty of the Music Production and Engineering Department at Berklee College of Music, Berklee president Lee Eliot Berk has announced. Szakmeister previously served as staff engineer at Newbury Sound Studios in Boston and Sound Loft Studios in Brookline, Massachusetts. He brings to Berklee an extensive background in multi-track recording and mixdown techniques.

• The recently-chartered **Audio Engineering Society Educational Foundation** has named three applicants as the first recipients of grants for graduate studies in audio engineering and related fields. In making the announcement, Emil Torick, president of the foundation, acknowledged the generosity of the benefactors whose contributions led to the establishment of the foundation. **James M. Mastracco** earned a B.S. degree in Physics at Union College in 1979, and an M.S. in Physics at Rensselaer Polytechnic Institute. He is presently a lecturer in physics and acoustics at Rensselaer and an acoustical consultant for the Troy Music Hall. He is a candidate for the Doctor of Engineering Science in Acoustics at Rensselaer and is engaged in concert hall acoustics research. **N. Charles Podaras** received his B.S.E.E. degree at Lehigh University in 1980 and the M.S.E.E. degree with emphasis on audio engineering and acoustics at Georgia Institute of Technology. Since 1981 he has been a member of the Technical Staff of Bell Laboratories. Podaras plans to continue his graduate studies in electrical engineering and digital audio signal processing at the Swiss Federal Institute of Technology in Zurich. **Anthony J. Romano** received a B.A. degree in Music Theory and Composition at Southern Illinois University-Carbondale in 1976. He then studied musical composition for two years under Olivier Messiaen at the Paris Conservatory, followed by further studies in recording techniques, sound synthesis and electro-acoustics at IRCAM, in Paris. He completed additional technical studies in physics and mathematics at the University of Illinois, Champaign-Urbana, and is presently enrolled in the Graduate Program in Acoustics at Pennsylvania State University. Grants for university graduate studies with emphasis on audio topics are awarded by the AES Educational Foundation annually.

• **KWMU-FM91**, the public radio station of the **University of Missouri-St. Louis**, recently finalized an agreement with **Audio and Design/ Calrec** that enables the station to begin regular ambisonic broadcasts. KWMU made history in May of this year by becoming the first station in the United States to air ambisonic programs and has continued to do so on a limited basis, KWMU becomes the first station in the country to broadcast regularly in the 2-channel UHJ ambisonic format. Ambisonics is a "surround sound" recording and broadcasting system that engulfs the listener in 360 degrees of sound, using sophisticated circuitry and a special microphone. The system's goal is to reproduce a performance with a level of realism unattainable previously.

• **National Semiconductor Corporation** has broken ground for a $75 million research and development center, adjacent to its corporate headquarters in Santa Clara, CA. When complete in the spring of 1985, the laboratory will employ more than 500 people. National Semiconductor is a leading supplier of semiconductor components and systems products. With 29 plants in nine countries, the company employs approximately 40,000 people worldwide.

• **Biamp Systems Inc.**, of Beaverton, Oregon, manufacturers of professional audio equipment, has announced the appointment of **Richard N. MacLeod** as company president. MacLeod, one of the original founders of Biamp, most recently served as vice president and Director of Engineering, with primary responsibility for the design and development of new products. Before co-founding Biamp in 1976, MacLeod was vice president of engineering at Sunn Musical Equipment, and previously worked as an engineer with Tektronix Inc.

SPARS Educational Programs

• The **Society of Professional Audio Recording Studios** has announced the establishment of three programs to provide assistance to audio engineering students. Each of the programs has been developed from pilot programs in operation during the past year. The SPARS Board of Directors has participated in interface days with audio engineering students at the University of Miami and the University of Colorado at Denver during the past year. The interface days provided an opportunity for students and faculty to discuss with active professionals the kind of preparation necessary for obtaining employment in the audio engineering field. Discussions covered a wide range of topics, including recent technological developments, the skills required in the industry, and the ever-changing nature of the recording studio business. SPARS plans to make interface days available to those schools interested in dialogue with the professional recording community. SPARS' president, Jerry Barnes of United Western Studios in L.A. noted SPARS' concern with providing the "real world" component of audio engineering education. The three-level program begins after the student's second full year of study, with a day spent in each of four or five professional studios. At the second level, after three years of study, the student will observe three studios for a period of three to four days. The purpose of the second level is to allow the student and the studio to determine a proper match for the third level, a 10 to 15 week working internship in one of the SPARS member studios. The third SPARS program will facilitate entry into the job market for the audio engineering graduate. SPARS will publish, twice a year, a resume book of those individuals seeking employment in the audio recording industry. The resume book will be distributed to SPARS studios and other studios that request a copy.

Consensus Reached on ¾ Inch Width For Digital Tape Recording

• The SMPTE Working Group on Digital TV Tape Recording (WG-DTTR) met at the Las Vegas Hilton Hotel last spring following the NAB Conference at which experimental digital television tape recorders were demonstrated by two equipment manufacturers. At a meeting of the Users' Subgroup, a consensus was reached that the 19 mm (¾ inch) tape widths should be recommended as the basis for a worldwide standard. The working group concurred and began preparation of a detailed list of data gathering experiments necessary to complete a 19 mm format specification. The goal of the SMPTE Working Group is to agree on a standard in cooperation with the relevant EBU Technical Group (MAGNUM), for consideration by the appropriate CCIR study groups. These CCIR study groups are responsible for developing recommendations for the international exchange of digital television programs on magnetic tape.

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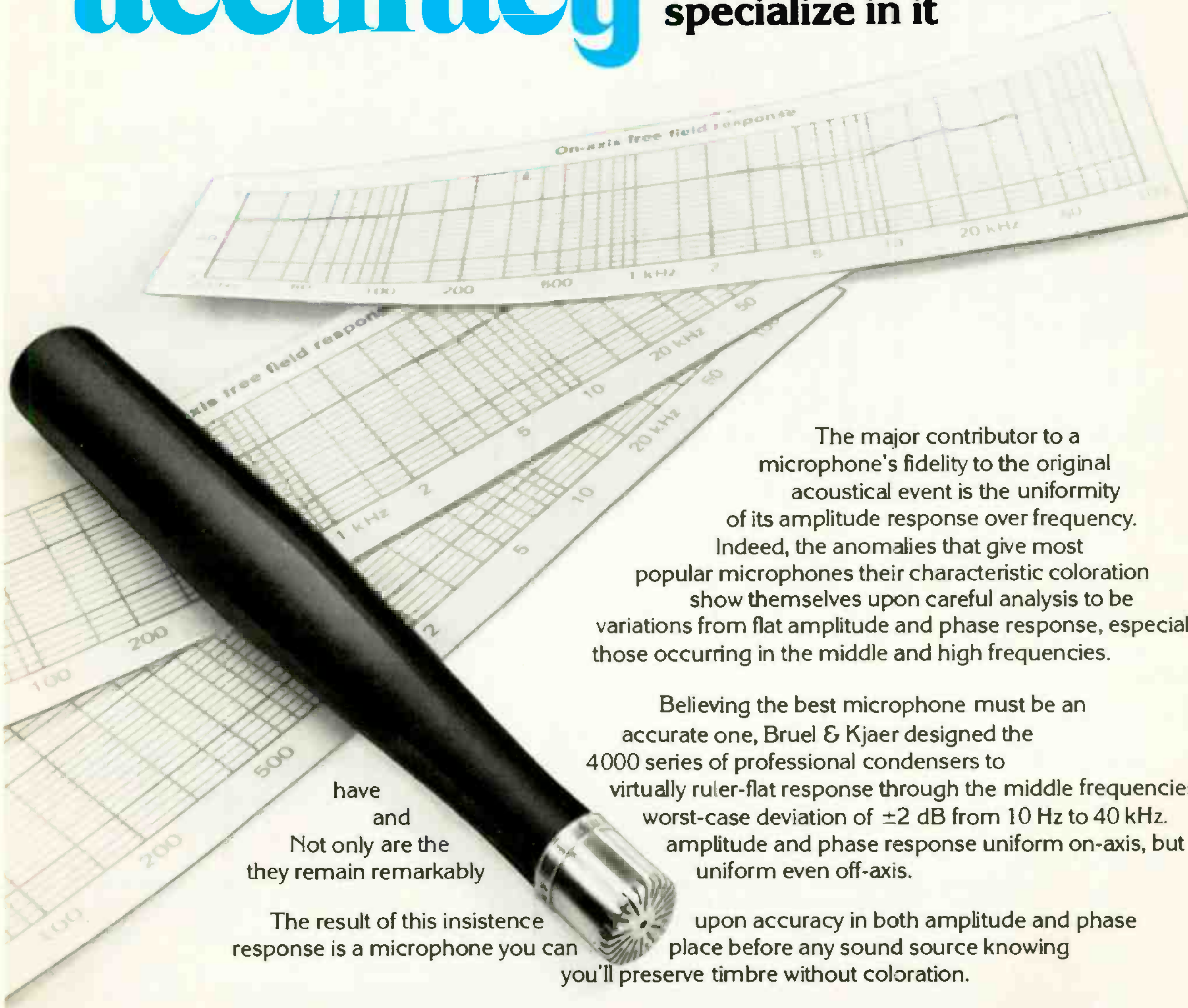

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