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Premiere Issue

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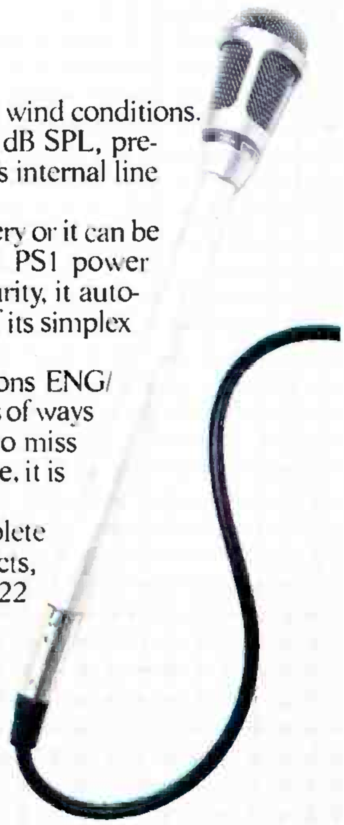

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NOVEMBER, DECEMBER 1983 VOLUME 17 NO. 10

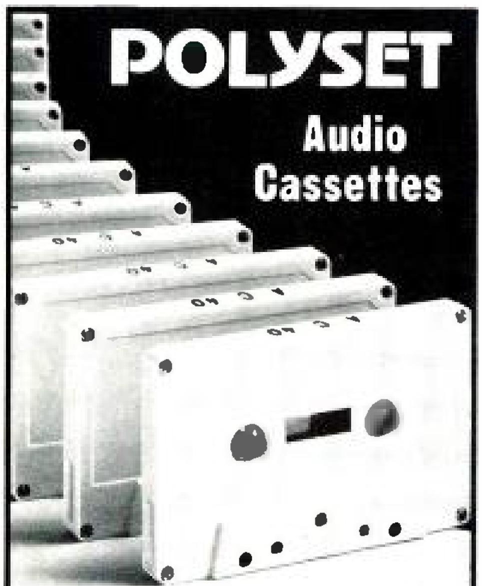
FEATURES

DEPARTMENTS

ABOUT THE COVER

• This month's cover celebrates db's 16th birthday, a birthday we're quite proud of. To mark this special occasion, we've reprinted two articles from the premier issue (along with 1983 updates), some "New Products" from 1967, and two ads from old friends Electro-Voice and Ampex that ran in our first issue.

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



Letters

STICKS AND SLATES

TO THE EDITOR:

Len Feldman's column "Sound With Images" in the July/August issue was very useful, but the reference at the end of it to the "sticks" technique as a method of synchronization is misleading.

The "sticks" or clap-stick slate provides only a start mark. The Nagra in question would have to be receiving a time reference signal for its neo-pilotone system with the same time base as the VCR. Then would come the classic situation on playback of matching the "getting-up to speed" time of each system, so that picture and sound could be in sync while running. It is done fairly often, but professional sync should lock to ±1/100 of a second, not the ± one frame in 16mm (36 feet/minute) standard we often hear about.

RICHARD FLOBERG
Associate Professor
Rochester Institute of Technology

TWO NEW INDUSTRIES ON THE WAY?

TO THE EDITOR:

I read with interest the editorial, "Quizzes and Exams" in the July/August issue of db. Mr. Boden's idea concerning formal education and the SPARS Certification Exam (SCE) is certainly meritorious and can hardly be argued against. More knowledgeable entry-level personnel are an asset to any industry. I would, however, warn you and Mr. Boden of a potential problem which may negate the function of your test.

In order to insure qualified and knowledgeable broadcast operators, the Federal Communications Commission administers a test leading to a General Radiotelephone Operator License. Many schools across the nation teach a course in FCC licensing. Students learn how to pass an exam and *not* broadcast engineering. They are able to pass the exam without *really* knowing much about current communications electronic technology. It makes sense that SCE courses will be offered that will produce people who can pass your exam and still know little

Index of Advertisers

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of audio engineering technology. This will depreciate the value of the SCE to the industry and render meaningless the scores of those who pass *and* really do know audio engineering.

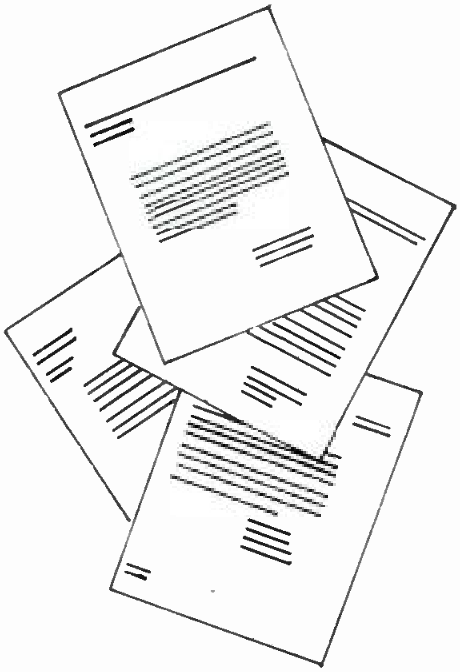
Secondly, the FCC exam is several decades old and does not reflect the current state of the art. It requires knowledge of obsolete technology and practices. With audio technology in such a rapid state of flux, the SCE could become obsolete in just a few years.

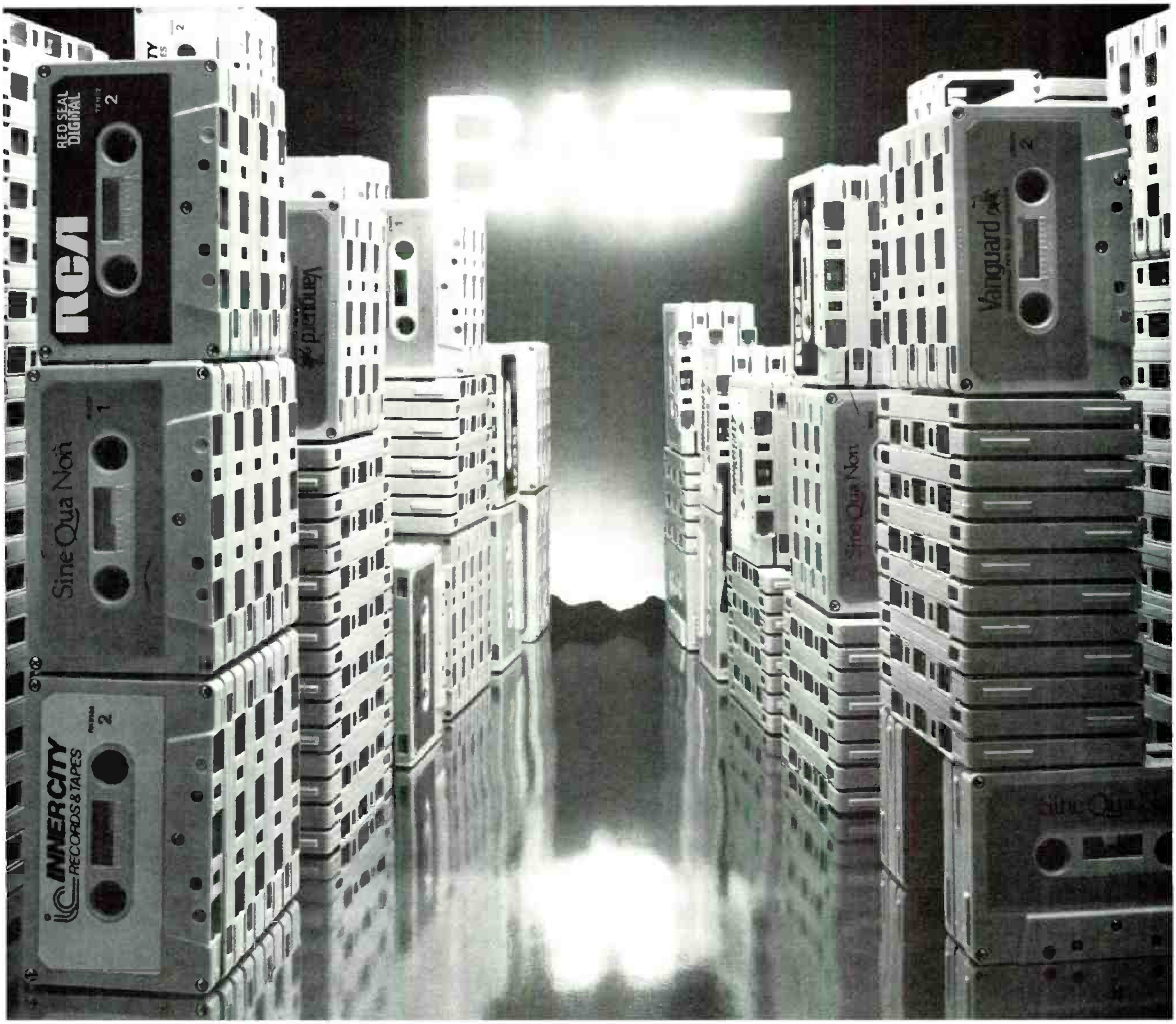
Before SPARS designs the SCE, I would hope that SPARS first solves the two potential problems just discussed. If they do not, all that will be "accomplished" will be the formation of two new industries: 1) the SCE course and, 2) the inevitable SCE exam manual.

RAPHAEL F. SEGURA

db replies:

Mr. Segura's point is well taken. But despite our fascination with audio, let's hope the industry is still small enough to elude the scrutiny of the "how to pass a test in 10 easy lessons" crowd.





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JOHN DELANTONI

The National Association of Professional Audio Manufacturers (NAPAM)

This month's contributor to "In My Opinion" is John Delantoni, president of Orban Associates, Inc. NAPAM is an idea whose time may have come. What's your opinion?

• NAPAM is the name I have given to a hypothetical for-profit association to be made up of manufacturers and distributors of professional audio equipment and ancillary services. NAPAM, which will be international in scope and nature, will be chartered to conduct group marketing activities (such as trade exhibits) and to lobby for more favorable rules and laws to enhance trade related to professional audio. The Board of Directors will be a seven-member unit composed strictly of people active in the manufacture and marketing of professional audio equipment for profit.

If the Audio Engineering Society permits, one member of NAPAM will sit as an ex-officio member of the AES Board of Directors (that is, with voice but not vote in Society business). There should, however, be no other interlock of officers.

NAPAM will present trade exhibits at such times and places and under such conditions as its membership permits by action of procedures outlined in its bylaws. The bylaws will also provide for modified parliamentary procedure with a greater than usual emphasis on membership voting and a hybrid, fast-track technique of conducting business via mail and telephone so as to prevent undue delay while retaining open communications among members.

In addition, NAPAM will be chartered to A) acquire revenue by its profit-making activities (e.g., trade exhibits), dues, and special assessments as proposed by its Board and ratified by the members at large and B) provide a stated percentage of its total revenue to the AES in recognition of its work in advancing the state of audio engineering and audio science, as a charitable contribution.

Membership requirements will include co-incident Sustaining Membership in the AES. The AES may wish to consider changing the structure of fees for any lost income not made up by the charitable contribution of NAPAM.

As a result of all this, NAPAM will permit the management of trade exhibits to more closely follow the interests of the commercial participants in audio engineering and alleviate the burden now shouldered by the Board of the AES—thus permitting a more effective and more responsive AES to further advance science and technology. In addition, income to the AES might well be enhanced and commerce vs. science bickering would be greatly reduced. ■

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RAISE YOUR STANDARDS.

That's why we designed the EVM PRO-LINE Series of extra heavy-duty sound reinforcement loudspeakers. The 15" and 18" models can handle 400 continuous watts of "real world" power (not just laboratory sine waves) and an incredible 1600 watts of peak music power. The 12" models will take 300 watts continuous and 1200 watts peak under the same grueling power tests.

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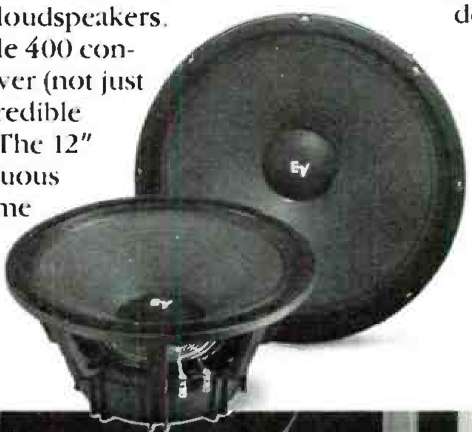
So raise your sound standards. See your EVM dealer or write for more information to: Greg Hockman, Director of Marketing, Music Products, Electro-Voice, Inc., 600 Cecil Street, Buchanan, MI 49107.



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Sound Reinforcement

JOHN EARGLE

Microphones in Sound Reinforcement, Part 2

• In most sound reinforcement systems, the microphone is located in the same spot that contains both audience and loudspeakers. As we all know, the gain of the complex loop shown in FIGURE 1 cannot simply be increased electrically without penalty. When the signal in the loop is amplified to a point at which the loudspeaker produces a level at the microphone equal to that of the talker, the system will "ring," or go into feedback. Actually, we cannot operate the system within about 6 dB of the ringing point if the system is to sound natural with speech input.

Fortunately, there are other system variables that can be manipulated to maximize the effectiveness of the reinforcement system. Even though we fix the electrical gain around the loop shown in FIGURE 1, the acoustical gain of the system need not be fixed. It is dependent on the distance from talker to microphone and, once we have fixed the coverage requirements of the loudspeaker array, this distance becomes our most important variable.

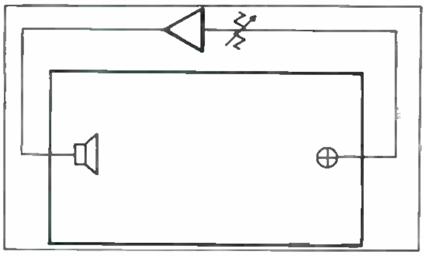


Figure 1. The electroacoustical feedback loop in a sound reinforcement system.

THE MICROPHONE'S DIRECT-TO-REVERBERANT RELATIONSHIPS

As a rule, the microphone is located within the reverberant field of the loudspeaker, as shown in FIGURE 2A. What this means is that the microphone is "acoustically" as far away from the loudspeaker as it can possibly be, since anywhere in the reverberant field the level from the loudspeaker will be consistent. (If the region in the neighborhood of the microphone is more absorptive than the rest of the room, some additional attenuation of the reverberant field may be expected, and it will increase the system gain potential accordingly.)

In normal use, the microphone will be well within the direct field of the talker, as shown in FIGURE 2B, but theatrical sound reinforcement may, through the relatively large talker-to-microphone distances involved, put the microphone in the transition region between the actors' direct and reverberant fields, as shown in FIGURE 2C.

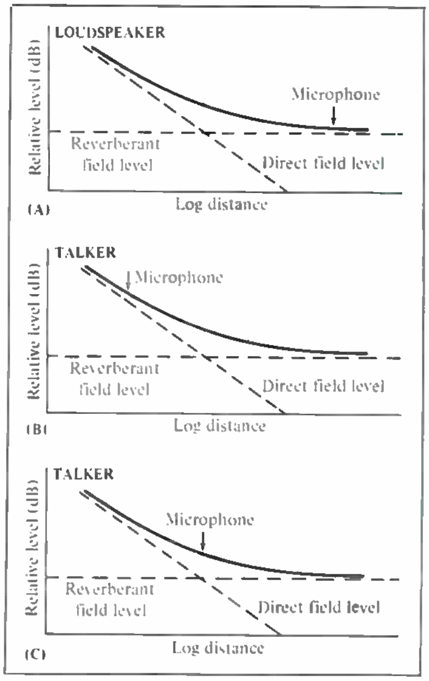


Figure 2. The microphone's direct-to-reverberant relationships.

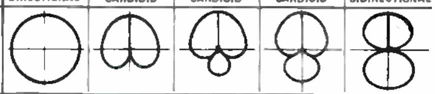
WHICH MICROPHONE PATTERNS ARE PREFERRED?

For the normal podium-mount application where there is sufficient acoustical gain, it is probably best to use an omnidirectional microphone because of its simplicity and well-behaved pattern. It is also free of proximity effect—which we will discuss in a later paragraph.

However, where there is a marginal gain situation, a directional microphone may offer some advantage. Referring to the data of FIGURE 3, note the next-to-last line in the chart, Random Energy Efficiency (REE). These values are given as numbers less than unity and also as negative levels in dB. Note that the omnidirectional microphone has an REE of unity (0 dB); this means that it picks up signals equally from all directions.

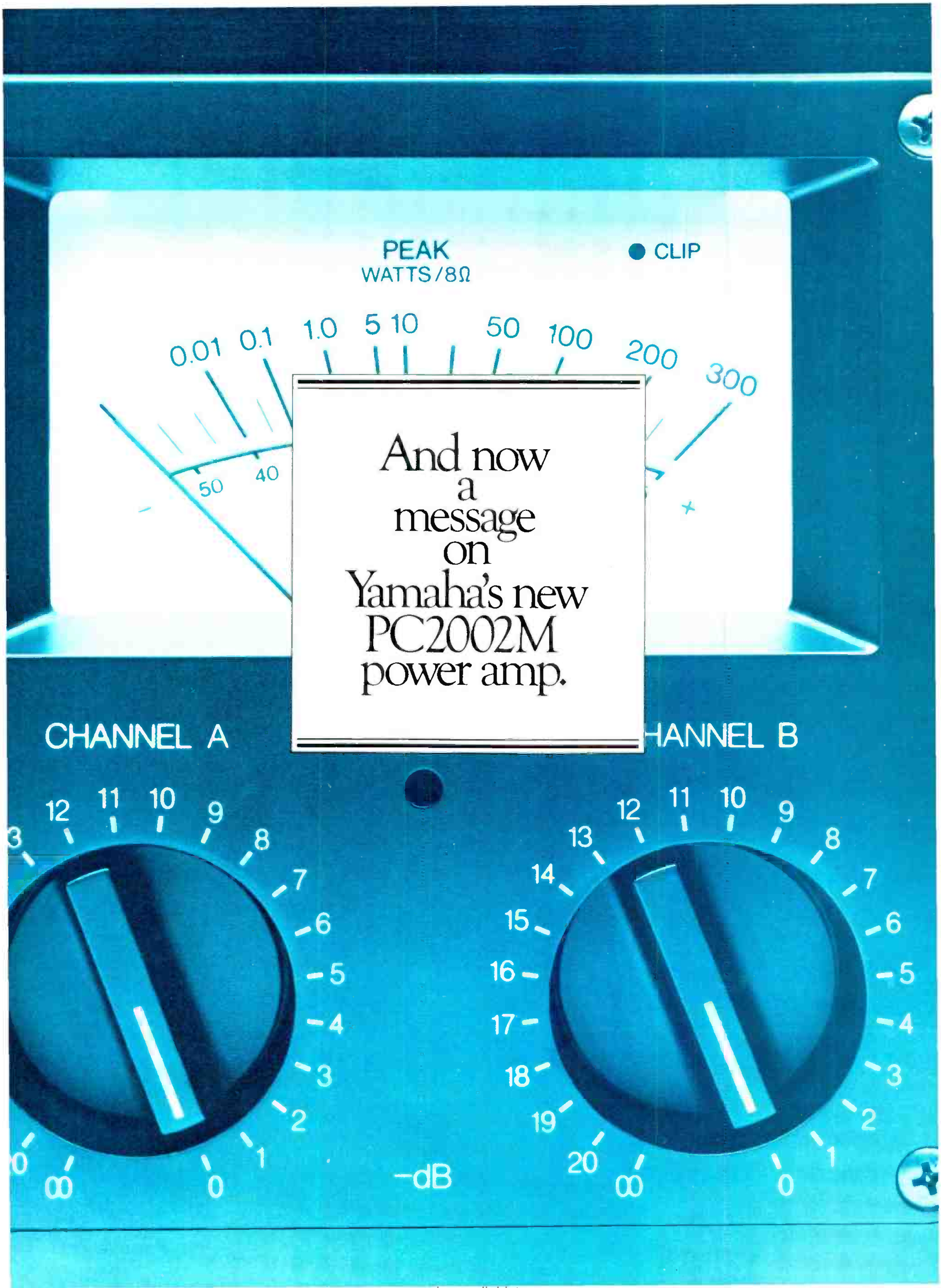
The cardioid pattern, with its REE of .333 (-4.8 dB), will pick up some 4.8 dB less reverberant information than the omnidirectional pattern when both are set for identical on-axis pickup of the talker, as shown in FIGURE 4.

Another way of looking at this is to consider the Distance Factor, shown in the last line of FIGURE 3. This indicates that, for the same contribution of reverberant (or random) information relative to the on-axis talker, the cardioid microphone may be placed at distance 1.7 times the distance of the omnidirectional microphone. The dis-

CHARACTERISTIC	OMNI-DIRECTIONAL	CARDIOID	SUPER-CARDIOID	HYPER-CARDIOID	BIDIRECTIONAL
Polar response pattern					
Polar equation	1	5 + .5 cos θ	.375 + 625 cos θ	.25 + .75 cos θ	cos θ
Pickup ARC 3 dB down (1)	—	131°	115°	105°	90°
Pickup ARC 6dB down	—	180°	156°	141°	120°
Relative output at 90° dB	0	−6	−8.6	−12	−∞
Relative output at 180° dB	0	−∞	−11.7	−6	0
Angle at which output = 0	—	180°	126°	110°	90°
Random energy efficiency (REE)	1 0dB	333 −4.8dB	.268 −5.7dB (2)	250 −6.0dB (3)	333 −4.8dB
Distance factor (DF)	1	1.7	1 9	2	1 7

NOTE
1 = Drawn shaded on polar pattern
2 = Maximum front-to total random energy efficiency for a first order cardioid
3 = Minimum random energy efficiency for a first order cardioid

Figure 3. A summary of first-order microphone data.


And now a message on Yamaha's new PC2002M power amp.
PEAK
WATTS/8Ω
CLIP
CHANNEL A
HANNEL B
–dB

PC2002M

SPECIFICATIONS

			PC2002/PC2002M
POWER OUTPUT LEVEL	Continuous average sine wave power with less than 0.05% THD. 20 Hz to 20 kHz	Stereo, 8 ohms Stereo, 4 ohms Mono, 16 ohms Mono, 8 ohms	240W + 240W 350W + 350W 480W 700W
FREQUENCY RESPONSE	10 Hz to 50 kHz, 8 ohms, 1W		+0dB −0.5dB
TOTAL HARMONIC DISTORTION	Stereo 8 ohms 120W Mono 16 ohms 240W Mono 8 ohms 350W	1 k Hz 20 to 20 kHz 20 to 20 kHz	Less than 0.003% Less than 0.007% Less than 0.01%
INTERMODULATION DISTORTION	70 Hz and 7 kHz mixed 4:1	Stereo 8 ohms, 120W Mono 16 ohms, 240W	Less than 0.01% Less than 0.01%
INPUT SENSITIVITY	Input level which produces 100W output into 8 ohms.		0 dB (0.775 V rms)
INPUT IMPEDANCE	Balanced and unbalanced inputs, maximum attenuator setting.		25 k ohms
8 OHM DAMPING FACTOR		1 kHz 20 to 20 kHz	Greater than 350 Greater than 200
S/N RATIO	Input shorted at 12.47 kHz Input shorted at IHFA		110dB 115dB
SLEW RATE		Stereo 8 ohms Mono 16 ohms	60V/µsec 90 V/µsec
CHANNEL SEPARATION	8 ohms 120W 8 ohms 120W	1 kHz 20 to 20 kHz	95dB 80dB
DIMENSIONS (W×D×H)	18-7/8×16-1/4×7-1/4" (480×413×183 mm)		
WEIGHT	PC2002 44 pounds (20 kg) PC2002M 45 pounds (20.5 kg)		

All specifications subject to change without notice.

The performance of the PC2002M speaks for itself. So does its sound, with exceptional low end response. And you can count on its superior performance over the long haul. We use massive side-mounted heat sinks, extensive convective cooling paths and heavy gauge steel, box-type chassis reinforced by heavy gauge aluminum braces and thick aluminum front panels. Yamaha's reliability is legendary, and with the PC2002M and PC2002 (same amp without meters), the legend lives on. For more complete information write: Yamaha International Corporation, P.O. Box 6600, Buena Park, CA 90622. In Canada, Yamaha Canada Music Ltd., 135 Milner Ave., Scarborough, Ont. M1S 3R1.

YAMAHA

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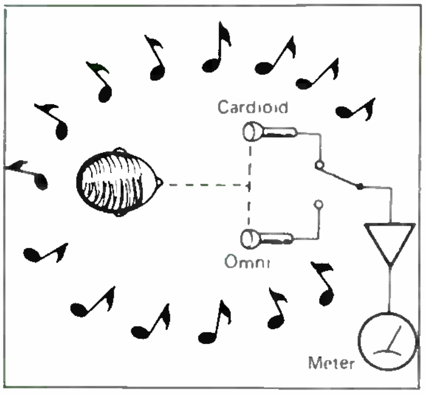

Figure 4. Random Energy Efficiency (REE). When both microphones are equidistant from a speaker, and are adjusted for equal level, the ambient noise level picked up by the cardioid will be one-third that of the omni (-4.8dB).

tance factor is then a measure of the "reach" of a directional microphone in just the same way that the directivity index of a loudspeaker is a measure of its "throw."

Note that the remaining cardioid variations, the supercardioid and the hypercardioid, have even more reach than the simple cardioid. Even though these patterns have rear-directed lobes, they have more overall rejection of randomly incident signals than the simple cardioid.

There remains a significant problem with the cardioid family of patterns—proximity effect. Proximity effect is the tendency of a directional pattern to overemphasize the low-frequency content of closely placed sources. FIGURE 5 shows the problem as typically observed. If proximity effect resulted in a fixed amount of boost, it could simply be filtered out; but that is not the case. The amount of low-frequency rise is inversely proportional to the talker-to-microphone distance, and any corrective filtering will only be accurate for one distance. Nevertheless, many directional microphones manufactured over the years have so-called "voice" and "music" switch positions that provide a rough compensation for proximity effect. Still other microphones, those that are intended only for close-up usage, are designed to be rolled off permanently at low frequencies, as shown in FIGURE 6.

The bidirectional pattern exhibits the greatest amount of proximity effect of all the first-order patterns shown in FIGURE 3, and for that reason alone it is not recommended for sound reinforcement work.

PICKING UP SOUND AT A DISTANCE

In typical legitimate theater speech reinforcement, microphones may be placed on the stage apron in the vicinity of the footlights. In those cases, the actors will be some distance from the microphones, and the system will not be capable of producing very much acoustical gain before ringing and coloration intrude. Fortunately, not much acoustical gain is required in such cases. More often than not, the purpose of a reinforcement system in the legitimate theater is to add just a little amount of high-frequency edge to improve articulation and to overcome high-frequency losses that may be inherent in the relatively absorptive environment of the theater.

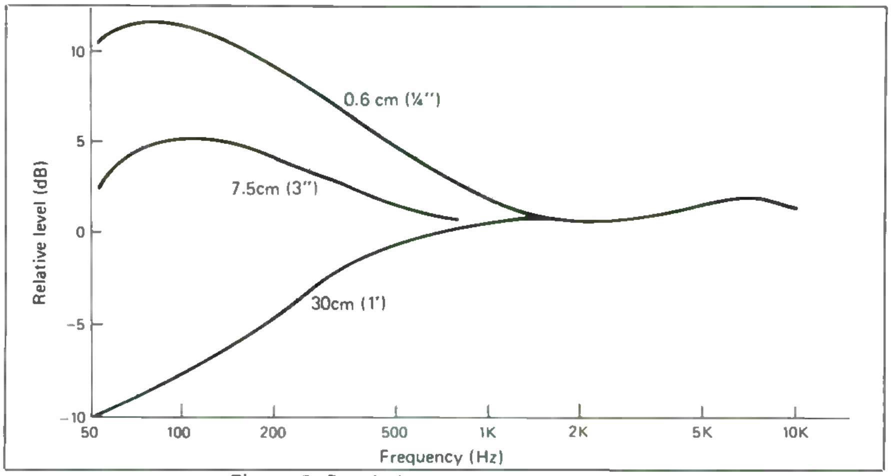

Figure 6. Proximity effects in a typical capacitor cardioid microphone.

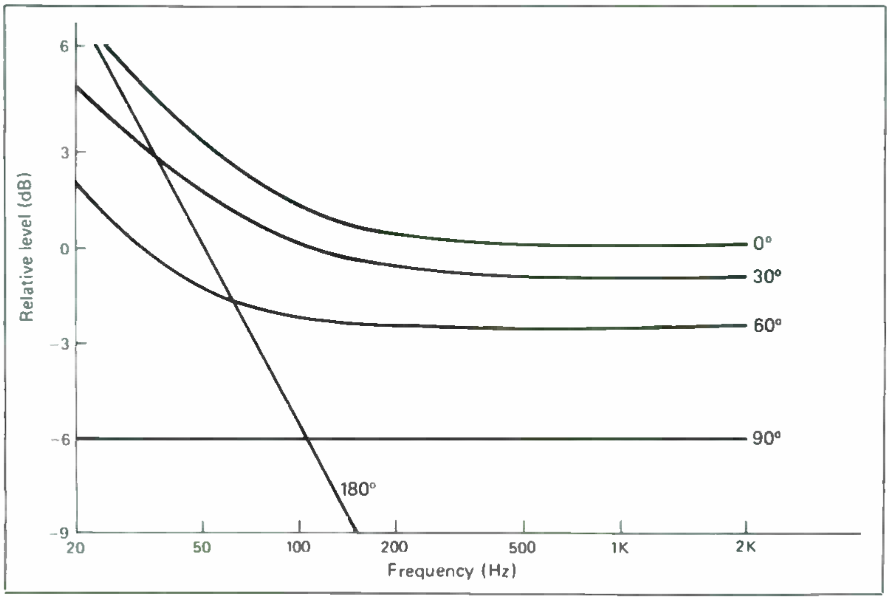

Figure 5. Calculated plots of proximity effect for a uni-directional microphone at 60 cm (2 ft.).

Almost invariably, omnidirectional microphones should be used for flush-mount applications, since any kind of directional model will see interferences from the close-by reflective floor plane. The most unobtrusive models are probably the Crown PZMs and other low-profile designs intended for surface mounting.

HIGHER ORDER MICROPHONES

The patterns shown in FIGURE 3 are all first order. What this means is that they are composed of simple constant and cosine terms, as indicated in the first line below the illustrations of the paterns. Higher order microphones are composed of squared cosine terms, and they are hard to realize physically. Their tight patterns are usually exhibited over a relatively narrow frequency range. They are difficult to use in most typical speech reinforcement applications, but they are useful in many broadcast applications.

Typical of these designs are the so-called "shot gun" microphones which, over the frequency range from about 700 Hz upward, exhibit distance factors up to 2.5. ■

KEN POHLMANN

The Other Digital Revolution

• I have recently heard some highly interesting comments about digital audio tape recorders and disc players. Certain individuals have felt compelled to come to the rescue of recorded music and save it from digital Armageddon, in much the same way that a number of years ago they probably came to the rescue of teeth when they were threatened by the Communist plot to fluoridate drinking water.

A LITTLE BACKGROUND

Reeves invented pulse code modulation back in 1939 and was granted a patent for it in 1942, while Shannon published his sampling and communications theories in 1948. But, apparently, some people only heard about (or bothered to listen to) digital audio this fall. Suddenly, digital audio detractors have been surfacing faster than pops on a vinyl L.P. Most of their statements are so convoluted that no known logic, metaphysical or otherwise, could ever disprove them. Listen to this: "the Compact Disc will not fool the ear forever," or, "as digital recordings enter the discography, there will be an evolution of decreased desire to listen to recorded music." As you can imagine, it is pretty difficult to respond to such statements. As far as their more vaguely specific criticisms, such as: "it sounds too cold," or, "it is unnaturally clean," they are equally hard to decipher. When digital manufacturers hear such things, and find themselves unable to meaningfully defend against meaninglessness, I'll bet they're filled with stress.

Anyway, some of the criticisms leveled against digital audio have been particularly nasty lately: I suspect the appearance of bins of CDs on the front aisle of most record stores has triggered the inevitable reaction from analog audiophiles. Frankly the vehemence of their attacks has indicated to me that they must feel that the end of analog is near. I can sympathize with their despair: things have come a long way since 1939 and a tremendous acceleration has occurred thanks to technology's development. Now that most leading studios have digital recorders (making digital recordings uneventful), and both digital players and digital recorders are entering the consumer marketplace, I can well understand their fear that the battle is almost lost. However, I would like to point out to them that it's even later than they think.

IS THE WAR OVER, ALREADY?

One digital critic has frankly admitted that for him all digital recordings have been disappointing, and that he simply cannot listen to digitized music. I feel very sorry for him because, soon, not only will all recorded music be denied him, but a lot of live music as well. His choice will be quite limited: all CDs—out; all digitally recorded vinyl—out; all analog recordings with digital effects or reverberation mixed in—out. Even if he goes to a live concert, he better bring his Digital Detector/Alarm for his well-being because he's not safe there, either. You see, as with all great revolutions, the digital audio revolution has consisted of warfare on several fronts, and while the recording community anxiously watched the headlines to see which studio would next go digital and at what sampling rate, it perhaps lost sight of the other digital revolution. More than recorders and players, discs and tape, more than reverberation and other signal processing, digital has revolutionized the very origins of music itself—musical instruments: that very tight drum sound—uh oh; that beautiful viola sound—uh oh! Musical instruments are getting digital, and in an especially insidious way.

THE COVERT REVOLUTION

The Synclavier II is a keyboard synthesizer; yet its manufacturer, New England Digital, likes to claim that it marks the end of synthesizers—not just the end of their competitor's synthesizers, but *all* synthesizers, including their own. Rather than joy in product mortality, their goal involves startling longevity. They envision an evolution of digital musical instruments in which they will grow to encompass the entire production process. They believe that the Synclavier II will grow to become a complete digital mastering studio—uh oh!

The Synclavier II is certainly a digital keyboard, and its ability is augmented by a 16-bit minicomputer with provision for floppy or hard disk storage. Anywhere from 8 to 128 voices or more (or partial timbres as they are more properly called), can be accessed from one key on the keyboard to create 96 harmonics; and using a chorus function, a key can trigger up to sixteen voices for 384 harmonics. A partial timbre contains the following: 24 separately adjustable harmonics, a six-stage volume envelope generator with control of delay, attack, peak, initial decay, sustain and final decay, a six-stage harmonic envelope for control of FM ratio, vibrato control with five waveforms and three waveform modifiers with speed, depth, and attack/decay time control, polyphonic portamento control with four variable rates, separate keyboard adjustment of envelopes for longer decay with lower pitches, chorus function for individual or all timbres, automatic repeat function, eight-assignment split keyboard, pitch control for different scales such as whole tone, quarter tone, microtonal, and scale variance to alter individual note values. Clearly, the Synclavier is a versatile keyboard ininstrument. But as a digital music system, we've only just begun.

The Synclavier II contains a 16-track digital memory recorder with punch in and out, fast forward and reverse (with a great inertia simulator), vary speed, vary pitch, time correction,

When you'd give an arm and a leg for an extra foot...

A-T gives you a hand!

The new AT835.

Now there's a new way to reach out and hear. The Audio-Technica AT835 Condenser Line + Gradient Microphone. It's barely longer than a legal pad, but it zeroes in on the sound you want to hear, while blocking out noise from the sides and rear.

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track time variation, track transposition, 16-track independent looping, track solo, pitch bending, click track, single track overdubbing, synchronization, bouncedown, timbre replacement, and, with the stereo option, dynamic panning is possible—as well as special effects. For examples, each partial timbre can be panned in one of 100 different phantom locations. The stereo option also allows the 16 tracks to be transferred directly to 2 track tape. The trick to all this is rather simple: the recorder is storing events, not the sounds themselves. The instrument's status is scanned every 5 milliseconds and position changes are recorded. It is, in effect, a kind of floppy disk-based automation package, minus the need for audio tracks. Okay, it's a great synthesizer with a high-powered sequencer and floppies. But there's more.

AND THERE'S MORE, AND MORE, AND...

The terminal option permits use of a video terminal such as the DEC VT100 to utilize the programmability of the computer. Three packages are available: a Graphics program, the Script music language, and Max, a programming language. Graphics is used to display timbral information either to analyze existing data, or to gain information to help create new timbres via the keyboard. Script can be used to input a score into the computer, accomplish editing, and then perform the piece in real time. Alternatively, a piece could be performed, and then its score created and displayed on the screen. In addition to standard musical notation, Script displays note lists for each of the 16 tracks, providing information on pitch, rhythm, articulation, timbre, and volume. Also, key signature can be altered, as can tempo changes, frames per second, special tunings, slurs and repeats. The Max program can be used to enhance the XPL structured language used in the Synclavier II and permits implementation of data structures different than the note-processing structure normally used. With a Z-80 CP/M option, the Synclavier II runs standard software and behaves like a personal minicomputer. Okay, okay, it's a computer, too. But there's more.

Sheet music printing is rapidly accomplished with Graphics, the printer option, and a printer. Standard music notation as displayed on the terminal CRT may be transferred to hard copy via a high resolution dot matrix printer. Since the system creates, stores, and performs compositions by itself, it's not clear who needs sheet music: can musicians even read music anymore? Speaking of non-conservatory types, an electric guitar option permits complete access to all of the Synclavier II's features: a 16-button LED panel attached to a Roland GR guitar controls real time features. Now the keyboard player, drummer, and guitar player are digital. Who's next, the bass player? And what about the lead singer? Uh oh!

As if all this wasn't enough, as if this hadn't carried musical instruments far enough into the digital domain, they had to invent the sample-to-disk option. Using a 5 MByte Winchester hard disk and 50 kHz 16-bit sampling and conversion, any sound or signal can be digitally recorded and fed into the system for performance or analysis. For example, a tympani is recorded/sampled and assigned to the keyboard for pitched playback: there is no need to carry those copper kettles around anymore; they have been reduced to bits. Or perhaps you want to create a new kind of tympani: use the sampled data to analyze the harmonic (or inharmonic structure) of a tympani (using an FFT or something), and in general play around with the bits to dream up something a little different, but based on authentic tympani acoustics. Although sample-to-disk is presently monophonic, a polyphonic version is currently in the works, first with 4 voices, then 8, 16, 32...the only limitation being available disk space. In other words, the instrument will have become a hard disk-based recording studio. At that point, the circle will have been closed and the war won —digital musical instrumnents and production and recording studios will have become synonymous hardware, differentiated only by the currently loaded software. The general purpose digital computer will have completely taken over the studio.

A LAST DIG

Well, I just wanted to have this little chat with critics of digital audio—just wanted to let you know that your forces have been flanked, and more or less encircled. You certainly were argumentative about digital storage methods. Your thesis that music was so incredibly complex that its information content defied digital techniques certainly bewildered a lot of people. Whereas everyone else in all other professions had gone digital, you stoutly maintained that music never could. Well, it sure was fun. But now that music itself is digital, because of the enthusiastic support of the composers who compose it and the musicians who play it, your obstinate role as a critical listener will just have to change.

While in the past it might have been interesting to protest against the digital recording of analog music, now anyone objecting to digitally recorded digital music would be considered a real crackpot. Uh-oh! ■

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BARRY BLESSER

Upconverting D/A

• One of the most novel applications of the re-sampling idea was presented to the Rye AES conference on digital audio by Plassche and Dijkmans of Philips Research Laboratories. They demonstrated how one can get better than 90 dB of signal-to-distortion ratio out of a 14-bit D/A converter. We all know that a theoretically perfect 14-bit converter would only yield about 85 dB, yet they correctly found a method to have a bit of a free lunch. They invented an old idea in a new format. They built a monolithic D/A using the re-sampling idea.

The key to the approach is that many D/A structures will actually convert at a faster rate than 50 kHz. They therefore took advantage of the extra speed to earn more dynamic range. We will develop the idea more extensively in order to understand how this works.

We start by observing that a perfect D/A will result in quantization noise that is uniformly spread over the allowable spectrum. The magnitude of the quantization error, or its energy, is determined only by the size of the LSB. The arguments were developed in the Digital Audio series several years ago but will be repeated briefly. Assume a 20 bit digital word which is a very accurate representation of the audio signal. If this feeds a perfect 14 bit D/A, then the last 6 bits do nothing since they have no corresponding bits in the D/A. The error is thus determined by the missing bits. If we define the LSB of the 14 bit device as being equal to 1 mV, then the remaining bits which we removed have a range of 0 to 1 mV. Since DC is usually not relevant, we say that the error is a peak ±0.5 mV. Since the worst case error is just as likely as any other error, we have to average the result. In technical terms we say that the error value has a uniform probability density function, since all values of the error are equally likely. With a little mathematics we arrive at an RMS value for the error which is given by

$$V_{err} = \frac{1}{\sqrt{12}} \text{ (LSB)}.$$

The next part of the argument is that these errors are independent from sample to sample. If the error on sample one is +0.22 LSB, then the error on sample two is just as likely to be any value over the range of +0.5 to -0.5 LSBs. This is the notion of statistical or uncorrelated independence. When this condition is true, the spectrum of the error is flat or white. There is an equal amount of noise energy in the region of 0 to 1 kHz as there is in the region of 1 to 2 kHz. Because the range of the spectrum is defined by the sampling frequency, we can determine the amount of noise in the audio band by simple division. With a 50 kHz sampling rate, the defined band is from 0 to 25 kHz. 80 percent of the noise power will be in the spectrum of 0 to 20 kHz. Using an analog meter, we would observe 1 dB less noise in the 0 to 20 kHz region than we would in the 0 to 25 kHz region. If the measurement had been made from 0 to 12.5 kHz, we would measure a noise level 3 dB smaller. In some sense, this shows the advantage of oversampling an audio signal.

In fact, delta modulation takes full advantage of this property since the effective sampling rate can be many megahertz. For each doubling of the sampling frequency, the noise is reduced by 3 dB in a *fixed* band. The total noise is constant but it is spread over a much larger spectrum. Mathematically, we can better describe this situation by creating a measure of the power per Hz. This is called the noise energy density. Sometimes it is represented as volts per root-Hz. The square root operation comes from the fact that noise voltage is the square root of noise power. To convert from noise density to noise, we multiply by the square root of the bandwidth if the density is expressed as volts/$\sqrt{\text{Hz}}$. Typical operational amplifiers are described in these terms.

In our discussion we have assumed that the noise density is spectrally flat. This is not always the case. With operational amplifiers, there is often a rise in the noise density at very low frequencies and a gradual rise at higher frequencies. With the quantization noise discussion, we start with the assumption that it is spectrally flat. This assumption is a good one when the signal is either high level and/or spectrally complex. The exact nature of the assumptions, however, is a difficult subject which is beyond the scope of this particular discussion.

OVERSAMPLING

How can we take advantage of the reduction in noise by oversampling? Think back to the discussion on re-sampling during the last few months.

The first part of a re-sampler is the increase in the sampling rate by injecting artificial 0s between true samples followed by a low pass filter. To increase the sampling rate by a factor of four, we add three 0s between each sample and then pass the resulting digital data into a lowpass filter which cuts off at 20 kHz but operates at an effective 200 kHz rate.

Although the original data might have only been 50 kHz, the new digital data is oversampled by a factor of four since it runs at 200 kHz. Conceptually, this is indistinguishable from having sampled the original audio signal at four times the rate. If the implementation of re-sampling uses high quality digital structures, the new signal has the same S/N as the original.

Now we are at a very interesting point. The new signal can be truncated, i.e., remove the lower bits, and feed to a D/A converter. The converter is running at four times the rate, so the quantization noise power is spread over a band which is four times larger. In the audio band from 0 to 20 kHz, the power density is reduced by four and the noise voltage is reduced by two. This gives us a 6 dB reduction in noise. The 14 bit converter now has a dynamic range equivalent to a 15 bit converter. The technique could be extended further by oversampling to 800 kHz. This would have produced another 6 dB reduction in noise. The ability to create fast data streams is a function of the hardware. At some point, the economics of this kind of hardware becomes unattractive. The digital lowpass filter becomes much more complex and much more expensive; also, the D/A converter no long is able to keep up with the data. There is clearly an optimum rate for oversampling, but this optimum is a strong function of current technology. Today's optimum will probably not be tomorrow's optimum.

NOISE SHAPE

The magnitude of the noise is determined only by the lack of low bits, but we can control the spectral shape using special techniques. Notice that, if we could change the shape of the noise spectrum but keep the magnitude constant, we could still further reduce that portion which appears in the audio band. Quantization noise no longer is representative of audio noise if we place the dominant part of that noise in the region above 20 kHz. Ordinary analog lowpass filters would then be able to remove it.

Spectral shaping of the noise is an old idea that has been applied to delta modulation systems and differential PCM systems. To appreciate the nature of noise shaping, we need to ask: Does the D/A conversion create the noise? We could answer yes, since the lowest levels are truncated. More appropriately, however, we should note that the act of truncation is actually performed before the D/A activity. The removal of the bits is done in the digital domain, not the analog domain. We thus need to consider a way of changing the signal before truncation such that the power in the truncated part has a different spectrum. The property of being spectrally white, in which the power is equal over the spectrum, comes from the fact that the error for each sample is independent. Could we make the errors dependent? The answer is yes. By selecting the nature of the dependence we can change the spectrum.

Let us illustrate the process with a DC signal. Suppose we wish to produce a DC signal of 1.03451 volts in a system which has the LSB defined as 1 mV. Such a system can produce a voltage of 1.0345 or 1.0356, but not the intermediate value of 1.03451. Now, let us consider a system which had nine samples at 1.0345 and one sample at 1.0346. The DC value is exactly 1.03451. This idea is identical to that of using dither. Notice that the digital input to the conversion process corresponds to 1.03451, and the error of 0.0001 is used to influence the main signal. After the errors have accumulated, the value present to the D/A is increased. In this kind of a system, the errors are not statistically independent. The 10th word of 1.0456 has an error of -0.0009 whereas the other nine samples had an error of 0.0001. These nine samples influence the error of the 10th. Such a system will not produce statistically independent errors and the spectrum will not be white. If we can find such an algorithm that reduces the voltage noise density in the 0 to 20 kHz spectrum by 6 dB, we would have earned another bit in the D/A process.

Noise shaping is a way of including the cumulative error over many samples. Normally, the error is only limited in its peak maximum. By accumulating the errors, however, the sum of the errors over many samples is further limited. Let us compare 16 samples at 200 kHz with error accumulation to four samples at 50 kHz without error accumulation. In the latter case, the sum of the errors will be 6 dB higher than the individual errors. (Statistical errors add in the square sense as if they were power.) The "average" error is reduced by a factor of four since there are four samples being considered. Relative to the LSB reference, the average is 0.5. Let us compare this to the accumulation case. Here, the sum of the errors cannot be greater than the peak because if it were, the accumulation would change the errors. The average is thus reduced by 16 instead of four. The above is only an approximation to the real case since we need complex mathematical equations to demonstrate the real operation of the system. We have only provided a plausibility argument.

Noise shaping does have some issues worth considering. The primary one is that of increased hardware cost. With the advent of VLSI (Very Large Scale Integration) it is possible to place this kind of complexity on a single monolithic IC.

There are several very nice by-products of the oversampling approach. The higher sampling frequency means that the anti-image filter is much easier to build in the analog domain. We are separating the 20 kHz region from the image at 180 kHz. This is a 900 percent difference, compared to the usual 25 percent difference. Because the filter is so simple, phase linearity is also possible; aperture correction is equally trivial.

All of these benefits come from the fact that the real burden of filtering is placed in the digital re-sampling filter. Since this has no drift or accuracy issues, once having been designed, a more uniform quality results.

ECONOMICS

The usual rules of economics dramatically change in this class of technology. With old-fashioned audio, we determine the cost on the basis of the number of components. With large scale integration, the circuit complexity itself is not a cost-related item. The economics come from other considerations such as size of the chip, yield in manufacture, development and testing. The size of the chip is important because it relates to the number of ICs that can be produced per manufacturing run. A run has a fixed cost. If we can make 1000 ICs from one wafer, the cost is 10 times less than if we can make 100 ICs. Similarly, the yield is very important. If 50 percent of the ICs are defective, then the manufacturing cost is (obviously) twice that of a 100 percent yield. Finally, the development and testing activity is so complex that the amortization is important. For this reason, VLSI can only really be considered for very large production runs, such as in the consumer industry.

For a given yield rate, it makes no difference what is on the IC. This makes monolithic ICs very interesting, even if there are complicated digital filters in the design. The oversampling D/A converter by Philips is a very good example of taking advantage of these issues. It is a mixture of analog and digital technologies in that it overcomes the limits in the analog domain by using digital means. ■

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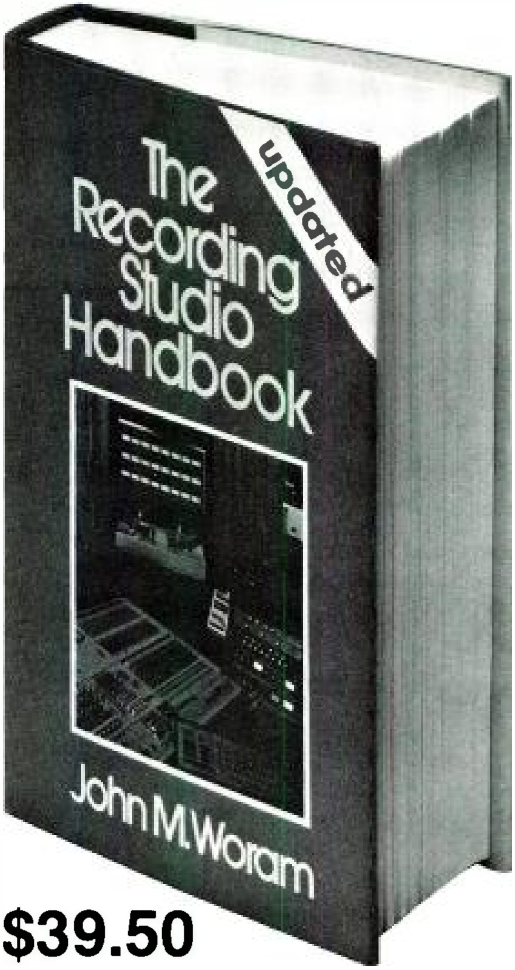

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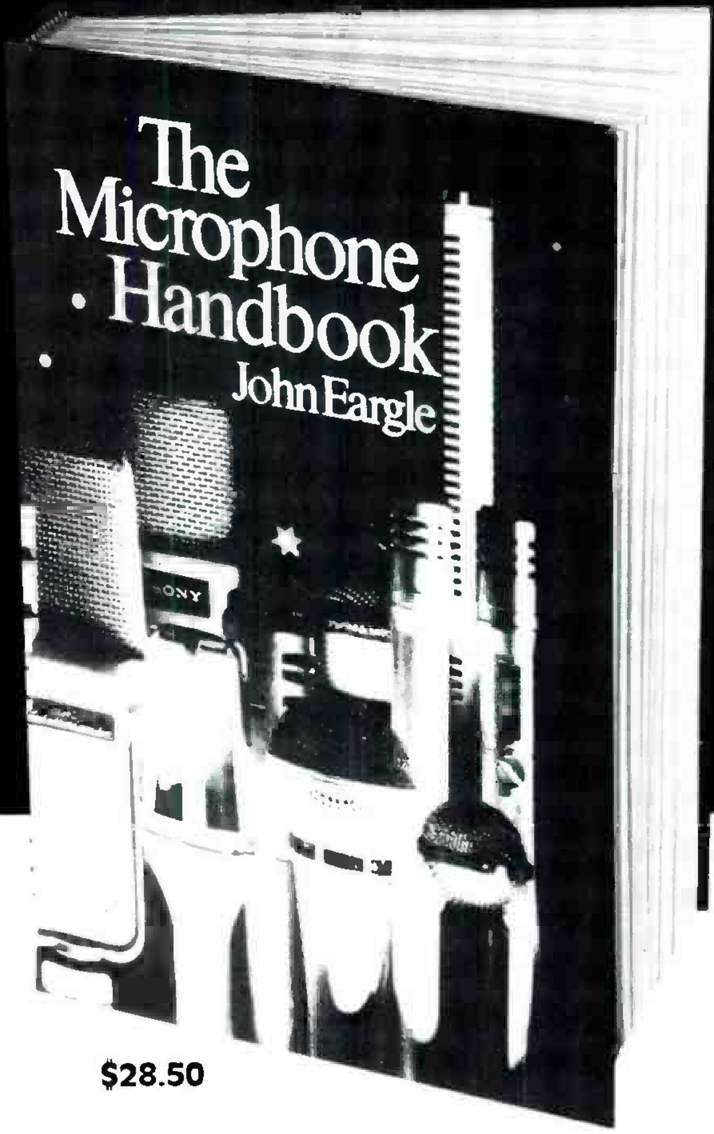

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Editorial Redux

"IN A FIELD as broad as professional audio has become, the dissemination of information necessary to the improvement of performance becomes vital. However, with the exception of the highly respected professional societies, no one has yet come forth to provide such service. **db**'s emphasis, therefore, will be to complement the theoretical orientation of existing publications with a focus on the practical aspect of new concepts. We will make **db** a central source for reports on new techniques as well as previews of new equipment and new ideas.

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What you have just read was written by me in 1967 as part of the first issue's Editorial. While it's true that other publications have come onto the scene since that time, it is also true that we have been faithful to our 1967 declaration for all those years and we will continue to be so. In fact, let me close with another quote from the first Editorial:

"We at **db** will endeavor to maintain a consistently high standard of technical accuracy and editorial excellence. We sincerely trust that our objectives will meet your needs and earn your encouragement."

We've been doing it, and we shall always try to do it. L.Z.

MARK McKENNA

Bearsville's Control Room B—Remodelling for the 80's

The remodelling of Bearsville's Control Room B revolved around the need for more space and a desire to be able to change the room's acoustics at will.

BEARSVILLE STUDIOS OPENED for business in 1970 as a 16-track studio catering to the needs of Bearsville Records and a surrounding musical community that included The Band. Todd Rundgren, Maria Muldaur, Paul Butterfield and many more too numerous to mention. The studio was built by owner Albert Grossman, who founded Bearsville Records after guiding the careers of Bob Dylan, Peter Paul and Mary, Janis Joplin, The Band and others. The premise of Bearsville Studios, then as now, was to provide a superior recording environment in a relaxed and beautiful setting.

Keeping pace with the developments in music recording requires almost constant research and revision, and Bearsville has gone through its share of changes. In the spring of 1982, chief engineer Ken McKim and then studio manager Griff McRee presented a comprehensive proposal delineating the need for an improved control room design for Studio B, one of Bearsville's two 24-track rooms.

Shortly thereafter, George Augspurger of Perception, Inc. was enlisted to handle the architectural and acoustical redesign of the room. Mr. Augspurger's reputation as a top designer in his field is built upon many successful room completions, as well as his tenure as a product manager with JBL. Initially, Augspurger developed four different layouts for the control room, tailoring them to the studio staff's request that the room be deader in the front and more live in the rear. McKim and staff busied themselves with the arduous task of selecting an appropriate console to replace the Bearsville modified Quad-Eight; the old desk had recorded gold and platinum records for the Isley Brothers, Meatloaf and Randy Vanwarmer, to name just a few.

OUTFITTING THE ROOM

After auditioning modules from several popular console manufacturers, an opportunity arose to acquire a Neve 8068

Control Room B featuring a Neve 8086 console, Studer A-80 VU MK III & 24-track tape machine, UREI Time-Aligned ™ monitors, and LinnDrum.

Mark McKenna is the studio manager of Bearsville Studios.

20 reasons why the QSC Model 1400 should cost more. And why it doesn't.

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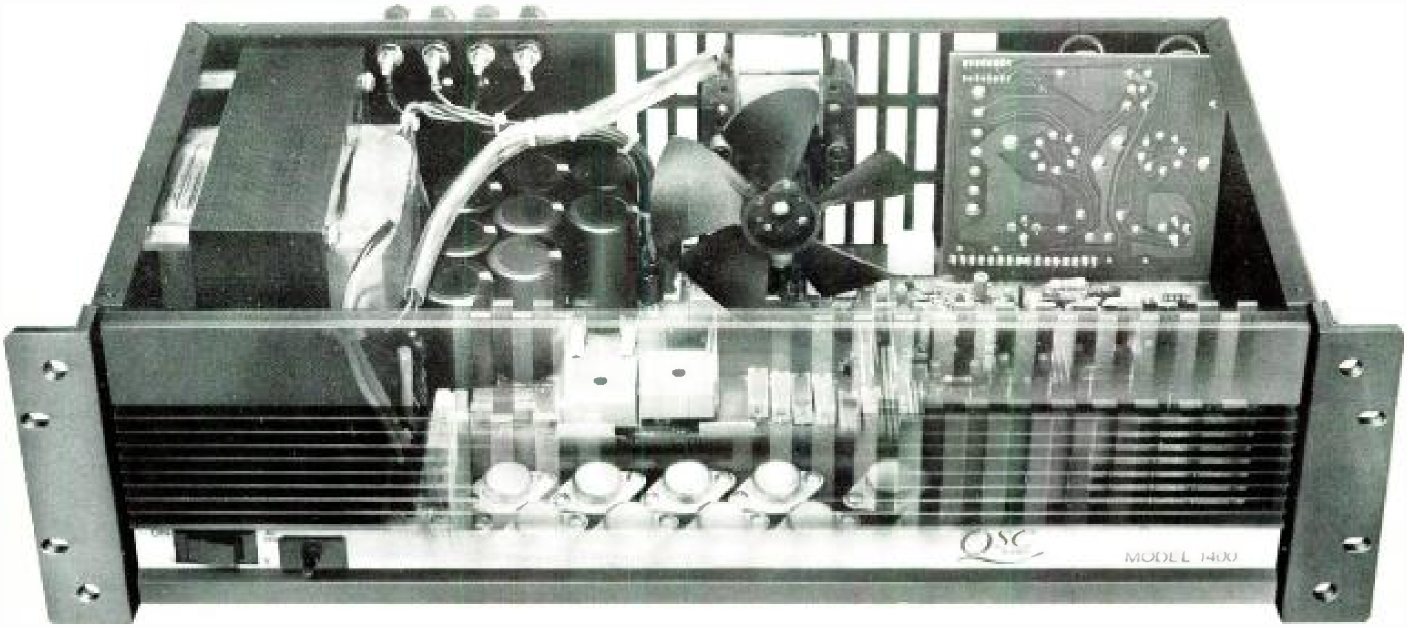

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in excellent condition. Soon after, Ken McKim flew to France to investigate the console, which was subsequently purchased by Bearsville. With its smooth sound and straightforward layout, the 8068 was a natural choice. Included with the console was the added bonus of eight Neve compressor/limiters.

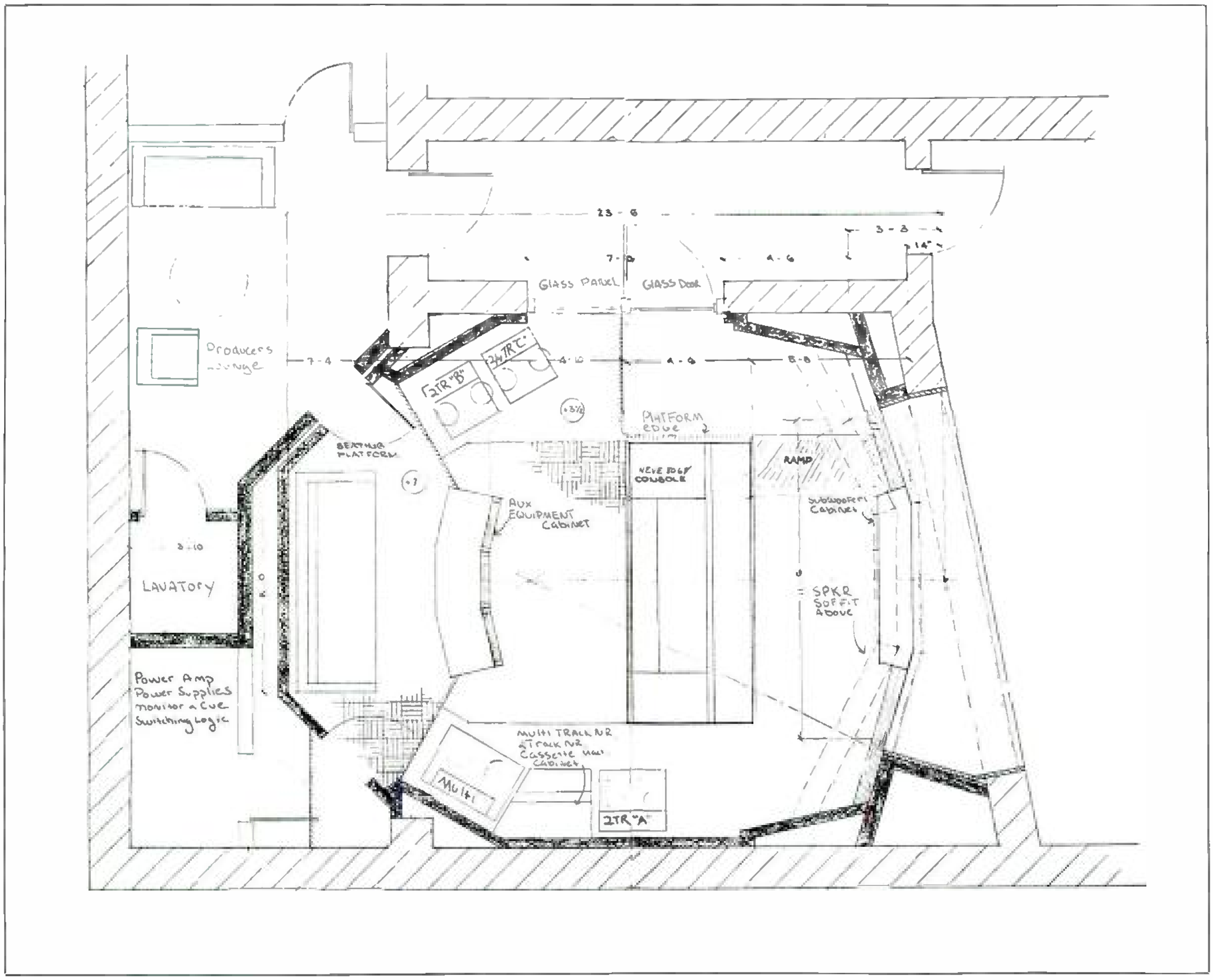
Since Studio B was to undergo a complete facelift, no aspect of the recording/monitoring chain was to escape review. McKim, McRee, and Grossman decided to install Studer tape machines in the new room: specifically an A-80 VU MK III 24-track, A-80 VU MK III half inch 2-track and a B-67 quarter inch 2-track. The decision was based on Studer's maintenance record, gentle tape handling and superior audio performance.

Selection of the control room monitors was based on the widespread acceptance of the UREI Time-Aligned (TM) Series. These speakers were first introduced to Bearsville in the Studio A control room, where a pair of UREI 811's, combined with a stereo subwoofer system, provided a pleasing sound with a "punchy" low end—and *no* room equalizers. George Augspurger is, of course, well known as a designer of studio monitors, but he commended the choice of UREI 813-As for Studio B because of their industry-wide appeal. His contribution to the monitoring system was the design of a stereo subwoofer system, installed in tuned ported cabinets under the control room window.

Meanwhile, Grossman and the studio staff had agreed upon one of Augspurger's room designs—with some revisions of their own. An addition to the studio building made it possible for Control Room B to absorb some space formerly occupied by a hallway, two restrooms and a janitor's closet. The old rear wall of the control room came out, allowing about ten more feet of depth. The added space made it possible to design a real "working" control room which could accommodate musicians who choose to play behind the console. Even with two or three people performing in the room, there is plenty of space for a producer, two engineers and a built-in couch in the rear of the room for guests. The slight elevation of this last area gives visitors a panoramic view of the activity. A low-slung outboard equipment cabinet that straddles the two elevations serves as a central workspace to accommodate a keyboard set-up or the like, and provides a natural barrier to keep the aforementioned guests from getting under foot!

REACH OUT AND CONSTRUCT SOMETHING

The construction of the new room was handled by Grossman's in-house crew of carpenters, headed by Paul Cypert.



Floor plan of Control Room B.

G. L. AUGSPURGER

New Acoustics at Bearsville B

WHEN A CLIENT decides to undertake major reconstruction of a studio or control room, the consultant's obligation is to analyze the constraints involved and decide whether the anticipated benefits justify the cost. In the case of Bearsville B, the decision was easy. By today's standards, the existing control room was too small and its geometry was awkward. Extending the control room into space behind would immediately provide a comfortable floor area and a well-proportioned volume to work in. I was satisfied that, one way or another, we could build a state-of-the-art control room in the space available.

I believe strongly that no design approach is the right one until others have been explored. In this instance, four floor plans were drawn up before we all agreed on the best approach. The final design is visually bilaterally symmetrical (and almost symmetrical acoustically), provides optimum geometry of console and monitor speakers, has room for a comfortable seating area behind the console, and sufficient space for adequate low frequency absorption.

As noted in an earlier article in db Magazine ("Contemporary Mixdown Room Design," Nov., 1981), non-symmetrical low frequency absorption can be a problem. In Bearsville B, however, the risk was not as great as it might appear. First, it's better to start out with too much trap area rather than too little; it would have been easy to later seal off part of the space had our guess proved incorrect. Second, the decision had already been made to augment very low frequencies with centrally located subwoofers, giving uniform left-right response below 60 Hz or so.

Q & A

Another design feature that may raise questions is the use of parallel side wall surfaces adjacent to the console. Not only are the surfaces parallel, but they are primarily reflective. Haven't we issued an invitation to the dreaded Standing Wave?

No, not really. The parallel surfaces exist simply to maximize usable floor space. If I had believed them to be acoustically unworkable they would not be there. Fortunately, I had sweated through three earlier acoustical corrections of rectangular control rooms and had some practical experience to fall back on. There are three good reasons why the Bearsville side walls do not introduce acoustical problems: (1) Parallel surfaces do not generate standing waves (standing waves are present no matter what the room shape), but they *may* set up flutter echo. (2) High frequency flutter echo is quite easy to avoid by introducing diffusion on one or both surfaces. Since the left wall is mostly glass, the right wall is highly diffusive. (3) First-order reflections from the monitor speakers miss the mixing area by a substantial margin. The reflective side walls provide higher-order reflections that are interpreted as desirable lateral diffusion of the late-arriving sound from either speaker.

WALL SURFACES

Probably the most interesting feature of the room, both visually and acoustically, is the use of vertical slat diffusers on all surfaces except the speaker wall. The variegated surface is highly dispersive at frequencies above 2 kHz or so. If the space immediately behind the slats is filled with fiberglass, then the surface in question is essentially absorptive below 2 kHz down to some low frequency cutoff determined by the depth of the cavity. However, if the surface immediately behind the slats is plywood or drywall, then good dispersion is achieved down into the mid-frequency range. Since the slats are built as plant-on modules roughly two feet wide, the acoustics of the room can be readily altered over wide limits without any change in appearance.

The adjustable wall surfaces are complemented by a three-tier "lily pad" ceiling. The front tier is acoustically absorptive and the remaining two tiers contain both absorptive and reflective surfaces. Although not as easy to change as the wall surfaces, these too can be re-configured if desired.

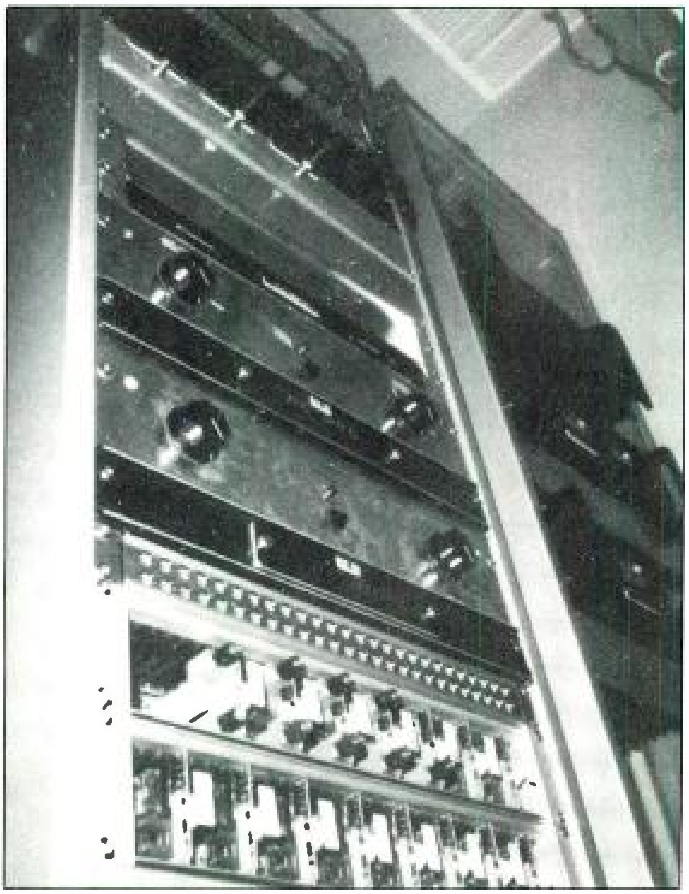
In our original design, about 25 percent of the room's surface area was absorptive. Acoustical tuning was done mostly on the basis of subjective judgment, since measurements indicated surprisingly smooth response from the start. We changed about 20 square feet of side wall treatment and then concentrated on surfaces and cavities immediately adjacent to the monitor speakers to zero-in on the best possible performance.

I was more than satisfied with the results. However, what pleases me may not please every client and every engineer. The beauty of this design approach is that it is easy to change the sound of the room without destroying it. Moreover, versatility has been achieved using common materials and standard construction techniques. ■

Since Augspurger's office is located in Los Angeles and Bearsville is in upstate New York, it was not possible for the designer to be present during all stages of the work. Augspurger sent plans for the room in sequence, and questions from Cypert and McKim were handled over the phone. When Mr. Augspurger arrived at Bearsville to see and hear the new installation, he was pleasantly surprised to find that it had been built exactly to his specifications. In fact, the actual construction came to within ½ of an inch of the original drawings.

The overview of the room plan called for structural symmetry and optimization of speaker geometry. This provides the engineer with excellent stereo imaging while including the entire console in the critical listening area.

Bearsville's "Assistant's panel," named because of its location on the assistant's desk to the right of the console



High-level room rack, featuring Crown, UREI and Yamaha power amps.

The monitoring environment seems bright, but not excessively dry or dead. This is due to the intelligent use of absorptive and reflective materials on the walls, ceiling, and floor. The slatted walls which line the perimeter of the room are configured to be reflective or absorptive in accordance with Augspurger's acoustical plan. Reflectivity is accomplished through the use of perforated Masonite panels; rigid fiberglass in a cut-out Masonite frame is used for absorptive areas. All of these Masonite panels are covered with colored fabric that dictates the color scheme of the room. The vertical wooden slats form the final layer of this wall panel "sandwich." The three rigid "clouds" which form the control room ceiling are similar to the walls in design, except that there are no wooden slats. The beauty of the system is that these treatments can easily be changed by removing the slatted wallpanels (they are Velcro'd into place) and substituting reflective or absorptive materials behind them. The room construction is, in a sense, modular.

ELECTRONICS

The electronic systems in the control room, from audio wiring to speaker muting logic, were designed in-house by Ken McKim and Raymond Niznik and are extremely flexible and complete. For example, there is a communication link with the studio that enables the engineer to speak to the headphones while simulataneously routing a mic located in the studio to the control room monitors. This mic can be defeated in favor of a monitor dim situation. The status selection for this system, as well as all the tape machine remotes, variable speed oscillators, reverb chamber decay controls, cue system on/off remotes, and Studer autolocator are contained in a custom enclosure known as "the assistants panel." This compact control center, designed by staff engineer/technician Niznik, neatly consolidates devices which are normally scattered over several locations in the console and outboard rack. It derives its name from its home—the assistant's desk on the right side of the console.

Another convenient bit of custom installation pertains to the studio LinnDrum which, when connected via its Bearsville DL interface, has all of its outputs routed to a patch bay adjacent to the mic preamp inputs. This arrangement eliminates the swarm of cables and direct boxes usually present during a LinnDrum session.

Out of sight, behind the rear wall of the control room, is the high-level room that houses the console power supplies, speaker and cue system switching logic and power amps: UREI 6500 for the 813-A's, Yamaha P2200's for the subwoofers and auxiliary speakers and Crown DC 300's for the cue system. Conspicuous by their absence are room equalizers. A thorough tuning of the UREI crossover networks and balancing of the subwoofer system during Augspurger's Real Time Analysis revealed a very smooth curve. There were none of the frequency response anomalies for which one-third or one-sixth octave equalizers are installed to correct. The resulting "straight line" signal flow from the console to the UREI 6500 amp provides a transient response that surely would have been blurred by even the best room equalizers.

THE VERDICT

The final assessment of any control room is how it rates sonically and how comfortable it is to work in. In both cases Bearsville's Control Room B is an unqualified success. Projects that have been mixed in B and mastered in a variety of disc-cutting suites have indicated that the room is accurate. The fact that musicians, producers, and engineers are able to withstand marathon sessions without contracting "studio burnout syndrome" attests to the room's attractive interior design (rendered by designer Bardet Storyk) and correct acoustic design principles. This combination of a first-class room with such top of the line equipment as the Neve console and Studer tape machines suggests the ultimate extension of an analog recording environment. ■

JIM WELLS

Audio/Musical Design for Animated Shows

The field of audio/musical design for animated shows is rapidly expanding, bringing with it new challenges for the audio engineer adventuresome enough to try his/her hand at it.

Part I: Visual Merchandising and Display or Teaching the Hamberger Elves to Sing

THE DESIGN OF AUDIO systems and the production of musical programs for animated shows and displays are interesting applications of the same basic tools and techniques used by engineers, producers, and designers in the pursuit of more conventional audio and music products. However, the demands made by the various "droids" with which we work often require a great deal of innovation and special thinking.

The animation audio/musical (A/M) designer must have a clear concept of all the variables that will contribute to the final listening experience. He or she must have a keen sense of how, when, and where to apply the basic principles of audio design, acoustics and studio techniques. Equally important, however, is a keen sense of how, when, and where to depart from standard practice, as each character and show will present its own unique challenges and special requirements.

By way of example we will examine some of the various engineering solutions necessary to the A/M designs in two individual, but as we will see, interrelated market applications.

The visual merchandising and display (VM&D) market presents extremely critical size, weight, set-up and budget limitations that force concept, design, and studio production innovation. These innovative solutions open the way for a new generation of audio/musical designs useful for the more traditional market areas of theme park attractions and themed entertainment centers.

THE BACKGROUND

When Fantasonics™ Engineering was commissioned to design an animated musical Christmas display that would perform continuous shows in a shopping mall center court, the idea seemed simple and straightforward. As we are primarily A/M designers, we began a research effort for both animation and animator. We wanted to find a VM&D animator whose mechanical figures were "ready-to-sing" with an on-board speaker and mouth animation, but we could not locate even one. As it turns out, characters with on-board speaker systems are unusual in any market, at any price (more about this in Part II).

However, after digging through piles of Christmas display catalogues and contacting every manufacturer we could locate, two things became evident: (1) everyone wants to do this—all of the display animation people were fascinated with the idea, many having pet audio projects "on the drawing board," and (2) no one as yet has done this—most of the excuses eventually boiled down to cost. Although we did find some efforts made with individual "talking" figures and some canned background music included with display, a musical program with multiple singing characters was considered cost prohibitive.

Indeed, most designers capable of creating an A/M animation are too expensive for all but the traditional mega-budget projects. The engineering efforts that have recently made animated shows financially accessible to restaurant/arcades are considered breakthroughs, and these shows can range well above the $100k mark. What the VM&D people don't know is that these costs are involved primarily in the creation of the animations themselves, their mechanics and control, and staging and set design.

We felt we could design an affordable show if we utilized existing animated figures, used the center court as the stage, and concentrated our R&D effort on an audio system design and music production. Certainly the universal interest demonstrated to us by literally everyone we spoke with justified a closer look. (Also, one should never use the word "can't" when in the business of translating fantasy into reality.)

THE PREREQUISITES

Upon closer examination, we found many engineering and design considerations that discouraged prior efforts. The following are only some of the problems that kept so many ideas "on the drawing board."

Budget. Budget is, of course, the major consideration. Although an A/M program greatly enhances the impact, effectiveness, and value of a display, our potential clients are working with fixed annual Christmas budgets on the average of $20k to $60k. This may seem like a great deal of money at first, but keep in mind that out of this amount must come all media buys and other advertising costs, Santa photo operation expenses, static and animated display pieces, hanging garland and lighting systems, set-up and tear down crews...and there's more. This most often leaves something on the order of a negative number left in the budget for the purchase of an audio system and music program; I exaggerate only slightly. But as will be seen later, we addressed these budget restrictions with a combination of

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

innovative system design and packaging, and a co-op marketing approach that spread studio production costs out over several systems.

Impact Life, Service Life. In all fairness to the manufacturers, many Christmas mechanicals enjoy a service life that spans several years. But in our target market, impact life of a display figure is purposely limited to a maximum of three seasons—or approximately three months actual use. For this reason, most figures are manufactured and priced to be practically disposable. Any on-board speaker system and mouth animation components must be engineered to be both reliable and effective and still remain inexpensive enough to complement the short impact life of the figure in which they are installed. In other words, any increase in figure cost must be held to an absolute minimum, and yet still sound good.

Logistics/Maintenance. For obvious reasons a shopping mall cannot afford an audio engineer to assemble and maintain a complex audio system. A major design criterion for any system is that it be quick, simple, and foolproof in its set-up and operation. Make that foolproof and bulletproof; the average mall's Christmas foot traffic can dismantle a tank if it is left parked too long in a center courtyard, and the average mall maintenance crew won't be able to reassemble it.

There is a limit to the amount of additional equipment that can be included with the display pieces. Too many components can create problems in set-up, storage, shipping costs and even cosmetic concealment. Many mall decor buyers are skeptical of anything mechanical, even animated figures. The recent proliferation of audio and computer systems seems to be dispelling most of these fears, but still, for all of the above reasons and more, simplicity is a key design consideration.

Sonic Impact/Acoustics. The primary function of any display is that of traffic builder. In order to generate the excitement and attendant increase in foot traffic that makes it cost-effective, the audio system in a point-of-purchase device must create a memorable listening experience for each individual shopper in a 360 degree viewing area. This 360 degree listening requisite renders conventional stereo and quad images awkward and unusable.

The institutional acoustics of most mall courtyards present several problems. We cannot risk unnecessarily exciting the characteristic long reverberation times of these large enclosed spaces. Our "minimum of additional components" must move a precise amount of air around the event in order to create an intimate performance that will draw traffic in toward it, and at the same time in no way interfere or compete with any adjacent mall function.

We must, however, optimize dynamic range and acoustic sound pressures, and pay careful attention to spatial and aural characteristics in creating a "live" quality to our droid's performance. We cannot expect anyone to believe our characters are truly alive; but with thoughtful design and production, we can sometimes make it easy to forget for a moment that they're just droids.

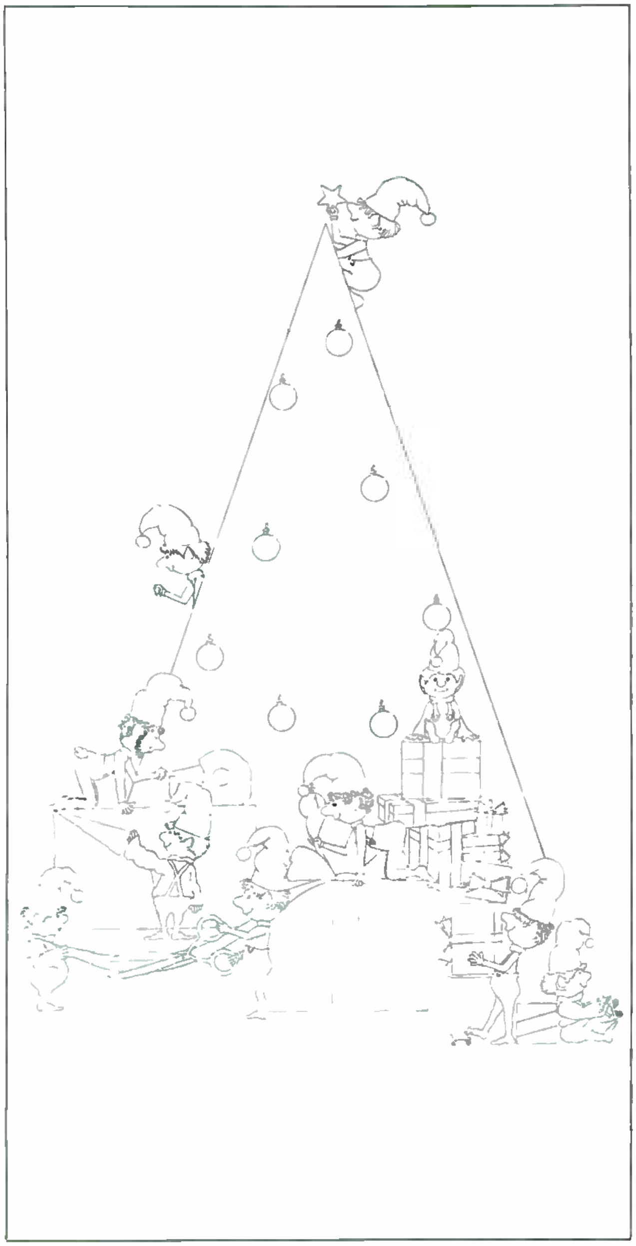


Figure 1. David Hamberger Inc.'s elves, featuring wrappers, trimmers (and gigglers?).

THE SIMPLE FACTS

Did I say simple? We simply have to create a design that is austere, lightweight, compact, virtually maintenance free, portable, disposable, able to be set up quickly by anyone, an acoustic event without being too loud, reliable, inexpensive, enchanting to everyone within earshot without giving any hint of the technical processes involved...and simple. (Whew!)

THE SHOW DESIGN

Our choice of animated figures was made easy when we found "the elves" offered by David Hamberger Inc. of New York (see FIGURE 1). These comical characters are engaged in the same holiday pursuits as the Christmas shoppers (who are the real elves!). They will (hopefully) encourage the viewer to laugh, and allowing us to laugh at ourselves painlessly is a primary function of all animated shows.

A musical concept breadboard was developed with a structure based on the symmetry of the audio system and display pieces. All of our elves (like the holiday shoppers) are "Getting Ready for Christmas," and this became the title of the composition. The elves were first divided into three activity groups: package wrappers, tree trimmers, and a group of smaller elves designated as gigglers and chucklers.

To facilitate on-board speaker loading, four figures were selected from each group and assigned to a discrete channel; the last channel(s) provide instrumental accompaniment to the music reproduction speakers. The gigglers are applied much like percussion instruments, but interact as part of the vocal arrangement.

A decision was made to keep the performance candid in that the elves (as all good elves should) will be singing as they work, not to the audience, but with each other. This posture helps to create a feeling of intimacy that encourages traffic participation and allows us to apply more subtle sound pressures.

A musical piece and lyric were written from the concept breadboard. The performance sequence begins with shimmering bells and windchimes; acapella tree trimmers are joined by various instruments as the music builds. Each group sings a verse and chorus about their particular activity, then both are repeated together for the finale. The piece subsides amid elfish giggles and laughter into a light percussion click-track waiting period. The entire cycle starts again every fifteen minutes without missing a beat.

The arrangement is charted primarily for string and woodwind quartets, with an emphasis on all types of bells in the rather elaborate percussion section. The other instruments in the arrangement include harp, wood recorders, synthesizer, acoustic guitar, and electric guitar with envelope follower. A small portion of the percussion section and the woodwinds were recorded first, performing to a click-track for the entire loop. The remaining instruments will

ARCHITECT'S SPECIFICATION

A short excursion down the signal path will help to illustrate the design solutions specified in this project. The extreme limitations in this market only serve to exaggerate the same basic design problems inherent in all animation efforts. The basic audio components also remain the same; it is in their application that the animation A/M designer must become a specialist (see FIGURE 2).

TRANSPORT SYSTEM

As of this writing, we are specifying the Compact Disc (CD) and Laser Disc player in almost all of our system designs. We chose the Sony CDP-101 as it offers many optional advantages over current playback-only "continuous use" tape transports and CD players alike. Most important to us is Sony's recent encouragement of commercial applications (FM broadcast, etc.). Development of a truly flexible professional unit with stacking capability and external microprocessor access will require Sony's help.

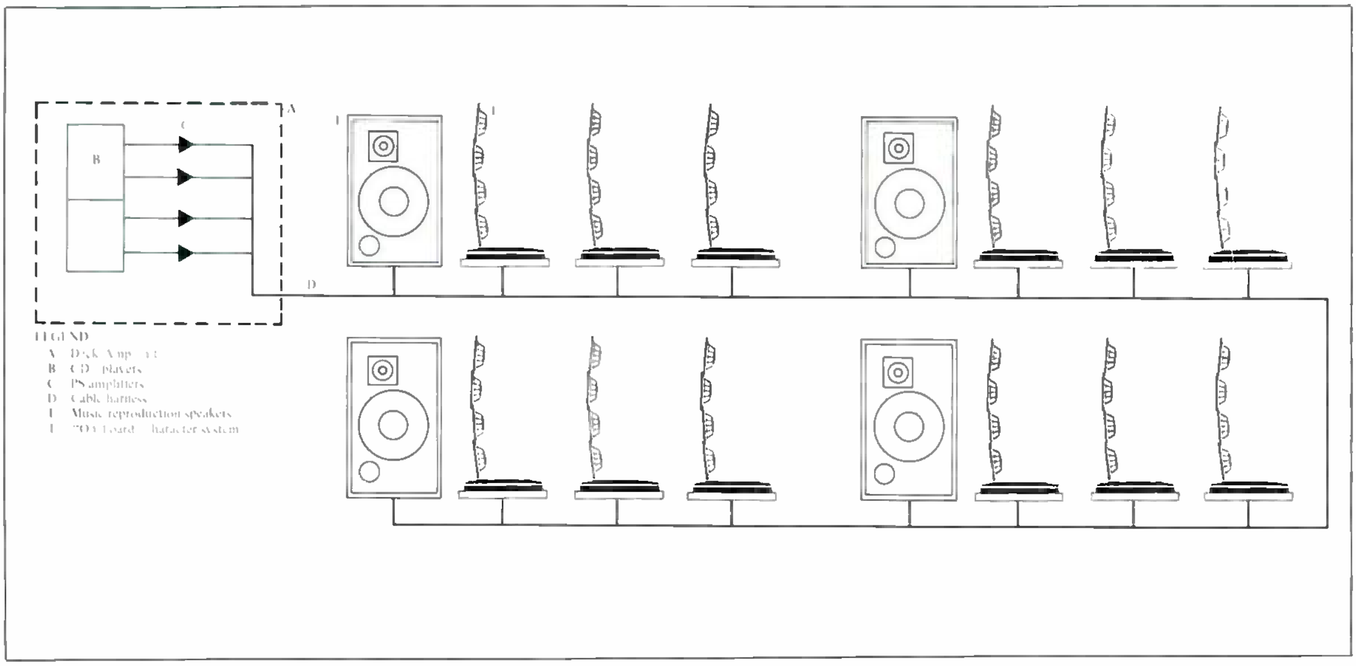


Figure 2. Block diagram of the visual merchandising and display (VM & D) design: A) deck/amp unit; B) CD players; C) PS amplifiers; D) cable harness; E) music reproduction speakers; F) on-board character system.

be recorded in real time with multi-tracking reserved primarily for voice work.

The "on-board speaker" system makes each figure its own character system and point source. A physical arrangement similar to that of a symphony orchestra or pipe organ is devised to make best use of sonic and visual movement within and around the 360 degree display.

The only component pieces necessary in addition to the on-board animated figure speaker systems are five Christmas gift boxes. These are a deck/amp unit and four music reproduction speakers. A single cable harness lies between the deck/amp unit and the various speaker systems. With the basic music program and audio design in place, work is begun on the architect's specifications and studio production schedule.

In the block diagram of our VM&D design (FIGURE 2), disc players (with minor modification) and tape players are interchangeable; in the acoustics of a mall courtyard, the difference is sonic predictability. Many of the "engineers" at Fantasonics are also "golden ears," and most of our multitrack work is still done in analog form. Digital mastering and the CD and player simply help to insure mix-down quality production "in the field."

AMPLIFICATION

For purely sonic reasons, we use the PS Audio Model II power amplifier exclusively in all our designs. The 2 × 2 power block configuration is an ideal package offering either two 70 watt (mono) or four 30 watt (stereo) channels in one 7½-in. rack space. These small amplifiers exhibit exceptional transient response and image fidelity due in part to a unique dual power supply design. Separate supplies prevent large demands in the output section from modulating a shortage of power to the front end. Exact reproduction of the information programmed onto each track is essential in conveying a live or lifelike quality, and these amps are transparent and natural sounding when confronted with digital dynamics.

CABLE HARNESS

The disc player outputs are each connected to one or more channels of amplification inside the deck/amp unit. The output of each amp is hardwired to a pair of pins on a single ITT K series multi-pin connector mounted near the base of the deck/amp unit. A mating connector is provided at one end of a 25 foot, 14 GA multi-pair spiral-rapped harness. Breakouts with phased connectors appear along the length of this harness, each marked for the various character and music reproduction speaker systems in physical sequence (see FIGURE 3). Once the display pieces are in position, complete system hook-up is accomplished in less than five minutes.

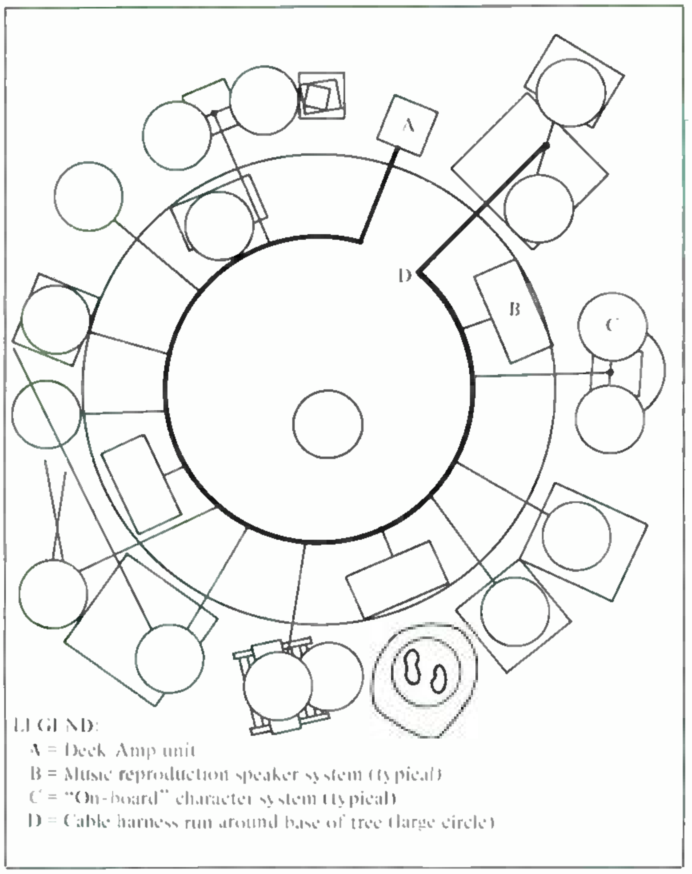


Figure 3. Overhead plan view: A) deck/amp unit; B) music reproduction speaker system; C) on-board character system; D) cable harness (runs around base of tree).

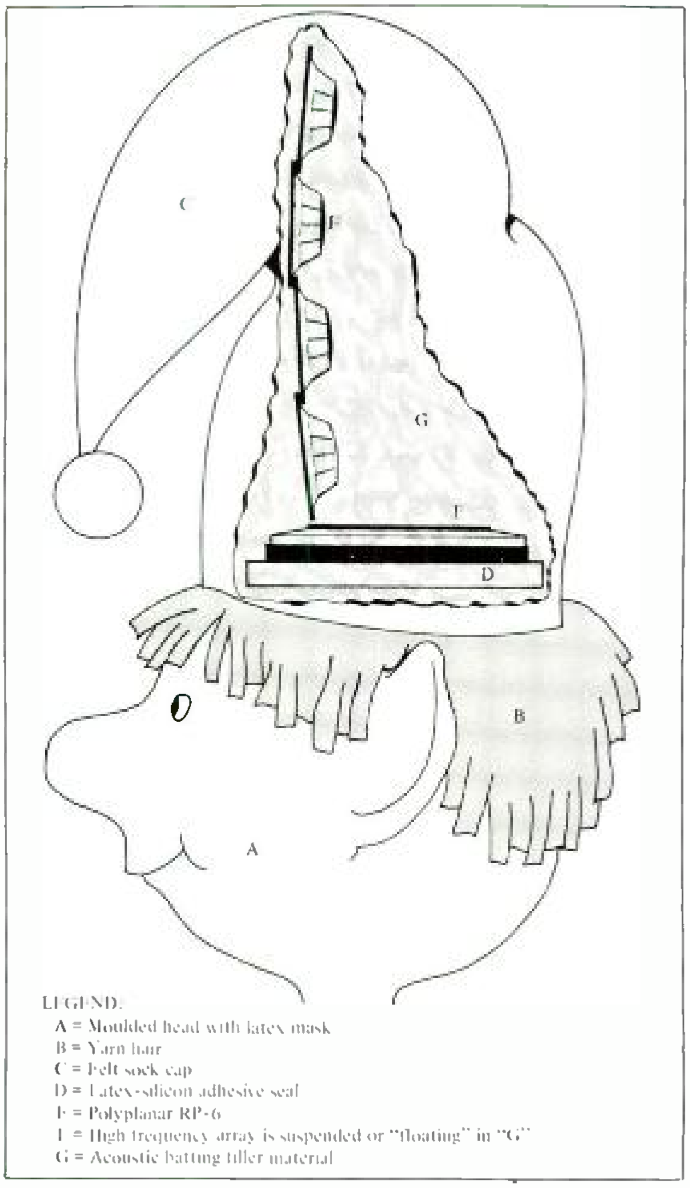


Figure 4. On-board speaker placement.

SPEAKER SYSTEMS

There are two types of speaker systems—an on-board character system and a music reproduction system. The latter systems are JBL 4301B broadcast monitors wrapped in colorful burlap and other acoustically transparent materials. These Christmas gift systems are placed either vertically or horizontally around the base of the display.

It is in the design of the character systems that we found the solutions to many problems. By incorporating speaker devices physically "on-board" the figures (as shown in FIGURE 4), we are able to eliminate any additional equipment beyond the five gift boxes and cable harness. Set-up of the twelve character speaker systems is accomplished by simply setting the animated figures in place.

Creating an on-board design creates its own problems. The components must be cosmetically concealed and the sock cap is the most likely location. However, mouth animation components must also ride in the head and there is a limit to the extra weight we can add and still expect the small electric animation motor to operate smoothly. Mouth animation is basically a circuit that samples the envelope of the signal going to the speaker and in turn triggers animation components. By including animation control on-board each character we eliminate the need for any computer control whatsoever. But this circuitry does contribute to the overall weight of the head.

We chose the polyplanar $5\frac{1}{4}$-in. speaker because of its compact size and weight ($5\frac{1}{4}$- × 1-in. at 170 grams). This speaker, when sealed to a hole in the top of the head cavity, works well in the voice range, but it radiates upward rather than forward. A special high frequency array employing four Motorola piezo ceramic devices was developed to focus directional characteristics forward, enhance point source enunciation, and reinforce sibilance and diction regions enough to project through a felt sock cap. This vertical line array creates the desired presence while contributing almost no additional weight (9.5 grams each).

Piezo ceramics (or PZT speakers) provide a two way system without the additional component cost, component weight, phase shift, signal loss and distortion of a passive crossover. These devices are capacitive by nature and roll in at slightly under 3 kHz, even when connected directly to a full range speaker system.[1]

The cost for the entire speaker system is under fifteen dollars, and total weight increase including the speaker system and the mouth animation components is only 265 grams! This low-cost, and practically weightless character system design allowed us to meet or exceed almost all criteria.

But wait, there is still the matter of sonics/acoustics. Our display must make a nice noise before we're done with it; that *is* the reason we are addressing all of these design problems in the first place. (Doctor, after the operation will I be able to sing?)

THE STUDIO PROCESS

The animation A/M designer's product is a Listening Experience. It is not just a studio recording or reproduction system, but the blending of these and other elements into the creation of a precise psychoacoustic perception in each listener. If an animated figure fell over in the woods and nobody was around to hear it, would it make any noise? I wouldn't debate the point, but nowhere is the answer more apparent than in the studio process.

A sonic model (see FIGURE 5) was developed and invited into the studio to undergo testing. When we heard the first voice signals reproduced by our prototype droid, we were somewhat less than enchanted. Response analysis revealed that the entire figure, especially the head cavity and latex moulding material, created severe aberrations in the overall response. Even with the voice pitch shifted slightly upward to create an elfish quality, the results were hollow and distorted. Unlike conventional designs, our "enclosure," the hollow moulded latex figure itself, adds a great deal of coloration. (Indeed, it is a latex Helmholtz Resonator!)

We want to create the illusion of a voice originated by the character, in a way that sounds natural. An elaborate patch hardly seems like the logical approach to natural sounding reproduction, yet through a complex chain of outboard signal processing we were able to optimize the signal to the speaker system and achieve natural sounding clean peaks in excess of 115 dB. The exact nature of this processing is considered a trade secret, but in essence we are compensating.

Much like the sculptor who starts with a 500 lb. block of granite, then chips away anything that doesn't look like a statue, we start with a recording and speaker system, then chip away at anything that doesn't sound like an elf. We want to precisely excite only desired aspects of the speaker/figure ensemble, and simultaneously reduce any characteristic of the recording process. We must use all means to minimize audible cues to the reproduction and recording processes if we are to create the illusion of a spontaneous acoustic event.

In reality, our speaker system or speaker machine is creating a spontaneous acoustic event. The voice reproduced by our elf bares little resemblance to the original performance as we heard it in the studio. Pitch shifting and the signal-speaker-figure interaction yield a voice that is altogether different, a voice that did not exist until our droid produced it! It is an incredible advantage in the studio process to know in advance the exact speaker system that will reproduce your product in the field.

A great deal of the technique applied to the voice recording was first accomplished in the testing and pre-production phases. It was during giggle pre-production work that we first experimented with pitch shifting to achieve an elfish or helium quality to the voices. On some occasions we actually did use helium (a small bottle of the compressed gas was brought into the studio especially for this purpose).

We employ two forms of pitch shifting in the giggle and vocal production. Tape speed pitch shifting is accomplished by slowing the tape speed several percent during recordings. This drops all tracks simultaneously; the vocalists simply perform in the new key. When the tape is returned to normal speed, the voices shift back up in sync and in key.

The second type of pitch modification employed is electronic pitch shifting. Slightly more involved, it creates a similar effect with a somewhat different character to it. In this technique the entire musical piece is performed on keyboard an appropriate interval lower, and recorded in sync on a vacant track. The voices to be digitally shifted are then recorded while listening only to the new track. These voices are bounced onto their final track assignments via an Eventide Harmonizer set to return them to the original key.

Two complete display figures with processing chains were kept in the studio for playback during these voice sessions; however, with the exception of pitch shifting, all signal processing remains outboard and is not actually recorded (or encoded) until the mix-down sessions.

THE "L.I.V.E." STUDIO MIX-DOWN

It was rapidly becoming evident that the special processing necessary to optimize each character system and the completely unique sonic image created by twelve individually animated point sources would require attendance of the *entire display in the studio* as a reference system during the mix-down phase. This is a variation of the L.I.V.E. (Listening Interpretation Via Event Environment) mix-down technique developed by Fantasonics™ Engineering for use in the more traditional animated shows (more about this in Part II).

The elves are referred to and even treated as performers, and will be kept in constant animation during the mixing sessions. An engineering approach similar to that used in concert reinforcement always proves to be quite helpful in giving a live performance quality to the presentation. All balancing, equalization, and other signal processing is applied as a result of direct interpretation of the animated display.

The most impressive characteristic of our on-board (or physically animated audio system) design by far, is the phenomenon created by separate moving point sources. If you can imagine twelve Leslie organ speakers all turning at slow speed, you can begin to get an idea of the effect produced. The physical animation of audio is almost hypnotic, and the choral enhancement created by multiple individual doppler shifts simultaneously must be experienced to be fully appreciated. This unique and fluid image has been nicknamed "The Calliope Effect," and walking around or past it is reminiscent of a ride on an old-fashioned merry-go-round.

There are literally hundreds of separate aspects and many months of work involved in a research and design project of this nature, much more than we have space for here. But by now it must have become evident that there is a great deal of ground-breaking involved in animation audio/musical design. Admittedly, this project kept us digging more than usual, but the resultant innovations unearthed are well worth any back pains. In Part II we will see how the physical animation of audio created by the on-board character system (patent pending) and the "Caliope effect" together with the live mix-down apply to a project in the more traditional market: the Bullwinkle Restaurant Show.

I hope one other thing has also become evident: dimensional animated audio/musical design is an exciting and challenging field with room for many new engineering, design, and recording opportunities. Anyone who feels the least bit creative and enjoys an occasional Chinese puzzle should look into this medium and all its potential. With animated shows appearing in more malls, arcades, pizza parlors, and theme parks every day, it is clear that this interesting field is limited only by imagination itself! ■

FOOTNOTES:

1. For more information on PZT speakers, contact: Roger Melewski or Sheryl Iverson, Motorola Ceramic Products Division, 4800 Richfield Road, NE, Albuquerque, NM 87113 (505) 822-8801.

JESSE KLAPHOLZ

Seeing What You Hear

With the help of FFT analysis, we are finally narrowing the information gap, and can thus begin to explore what is actually happening in real situations.

FOR YEARS I HAVE been hearing from so many different people in the music and sound business: "Yes, the RTA says it's flat, but that's not what it sounds like" or "Let's pink noise the system first, and then we'll listen to the last Brecker Brothers Band LP and fix it up with EQ." The crux of the problem has been an "information gap"; we simply have not been able to "see" what it is we are hearing and dealing with. The goal of this article is to take a look at how engineers and sound operators have dealt with systems in the past, the problems encountered, and what this author did about the information gap.

The problem begins with the fact that this is not only a highly scientific field but an artistic one as well. There has to be a constant balance between science and aesthetics for any project to achieve its goal. Too often after I have given a long dissertation on how "you can't cheat the laws of physics," my listeners have responded, "OK, but what about the real world?".

Prior to Altec's Don Davis and Arthur C. Davis developing the 1/3 octave equalizer in 1967 and Hewlett Packard's introduction of the first 1/3 octave real time analyzer in 1968, equalization was an incredibly tedious task which could and often did go on for days! With no standard to go by, there were many different types of chart level recorder systems being used for measurement and analysis. With the advent of 1/3 octave equalizers and RTAs, a system could be equalized within a relatively short time frame.

Merrily we EQ'd with our pink noise, RTAs, and 1/3 octave equalizers, but when the curtain opened we said to ourselves, "Oh my God that sounds horrible." Even with the house curves and all sorts of mic tricks, it was to no avail; it still sounded different with music. So what did we do after we ironed out all the ground problems, polarities, electrical gain, alignment, proper coverage, level, s/n ratio, etc.? We listened to the system. Could it be that we could hear better than the RTAs we just shelled out all those bucks for?

After using 1/3 octave analysis and the UREI X-Y plotter system, I was still not satisfied with what I was seeing and hearing. Out of the blue one day an ad appeared in the back of the AES journal for an inexpensive FFT analyzer for the Apple II computer. About fifteen minutes elapsed befor I ordered the owner's manual. After three hours of reading the manual and $28.74 of Ma Bell's "Reach Out and Touch Someone," the IQS 401-L FFT Analyzer for the Apple II computer was on its way. Then I realized, I better run out and buy an Apple before my IQS arrives!

It took some time before I was really able to get around on the computer keyboard and manipulate this incredibly powerful machine. It was like the time a friend of mine let me drive his $38K Mercedes: awesome! For the first time, when someone subjectively described a physical phenomenon to me, I was able to actually see what that person was physiologically hearing!

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

THE FFT ANALYSIS SYSTEM

The IQS 401-1 FFT Analyzer is a hardware and comprehensive software package designed for the Apple II personal computer. (The basics of FFT analysis were covered by Robert Berkovitz in the August and September 1982 issues of db).

The basic system architecture (1) is outlined in FIGURE 1. The preamp (FIGURE 1B) is gain-controlled by a DAC used as a digitally programmed attenuator, via software operation. In the feedforward compensation mode, the preamp has an extended gain bandwidth factor of ±0.2 db from 5 Hz to 40 kHz.

The next stage consists of the anti-alias low pass filter, which is programmable in order to accommodate multiple sampling rates, and has a 100 dB/octave rolloff. The output of the filter routes to the sample and hold amplifier, where digital data is strobed out onto the data bus via a DAC.

The test signal generator circuit consists of a DAC that can receive instructions directly from software. This allows simple software driver routines to create a wide variety of waveforms for many applications. For example, it can be used to provide impulses that are software controllable in amplitude and width for impulse testing. Another application is to "play back" waveforms at any speed from memory which may have been previously sampled, brought off disc, or synthesized by inverse FFT. Such waveforms could consist of gated sine tone bursts, pre-recorded sounds, sections of speech, etc.

Referring to the system block diagram in FIGURE 1, the digital portion is represented by the lower 2/3 of the diagram. It provides interface, memory, timing, and logic control functions. The SYSTEM CONTROL LOGIC block determines how each module interacts with one another and links the overall system to the Apple II bus structure. In order to maximize efficiency, many of the most often used machine code subroutines (including the FFT program) are placed in read only memory (ROM). Three ROM sockets (ROM BANK 1-3) are incorporated in the hardware, each providing 4 kilobytes of program storage. A single 4-kilobyte ROM is installed in the present version, thus allowing 8 kilobytes of expansion for additional signal processing functions.

THE SOFTWARE

Written in BASIC, the control program can be easily modified (or a new version written from scratch) to accomplish specialized tasks. For instance, establishing whether or not a given transducer conforms to an accepted standard curve might require a sequence of steps from the original control program. Maximizing the use of software plays a major role in the control of system costs; this is one feature that makes the IQS an economical package.

There are two basic modes of the control program, the ACQUISITION MODE and the ANALYSIS MODE. All of the commands are invoked by a single keystroke, such as A = AMPLIFICATION, V = VOLUME, P = PLOT, etc. Pressing a single key such as "S" would bring up the "sampling rate menu" onto the screen. There are eight different choices of sampling rates available; these are shown in FIGURE 2. When the

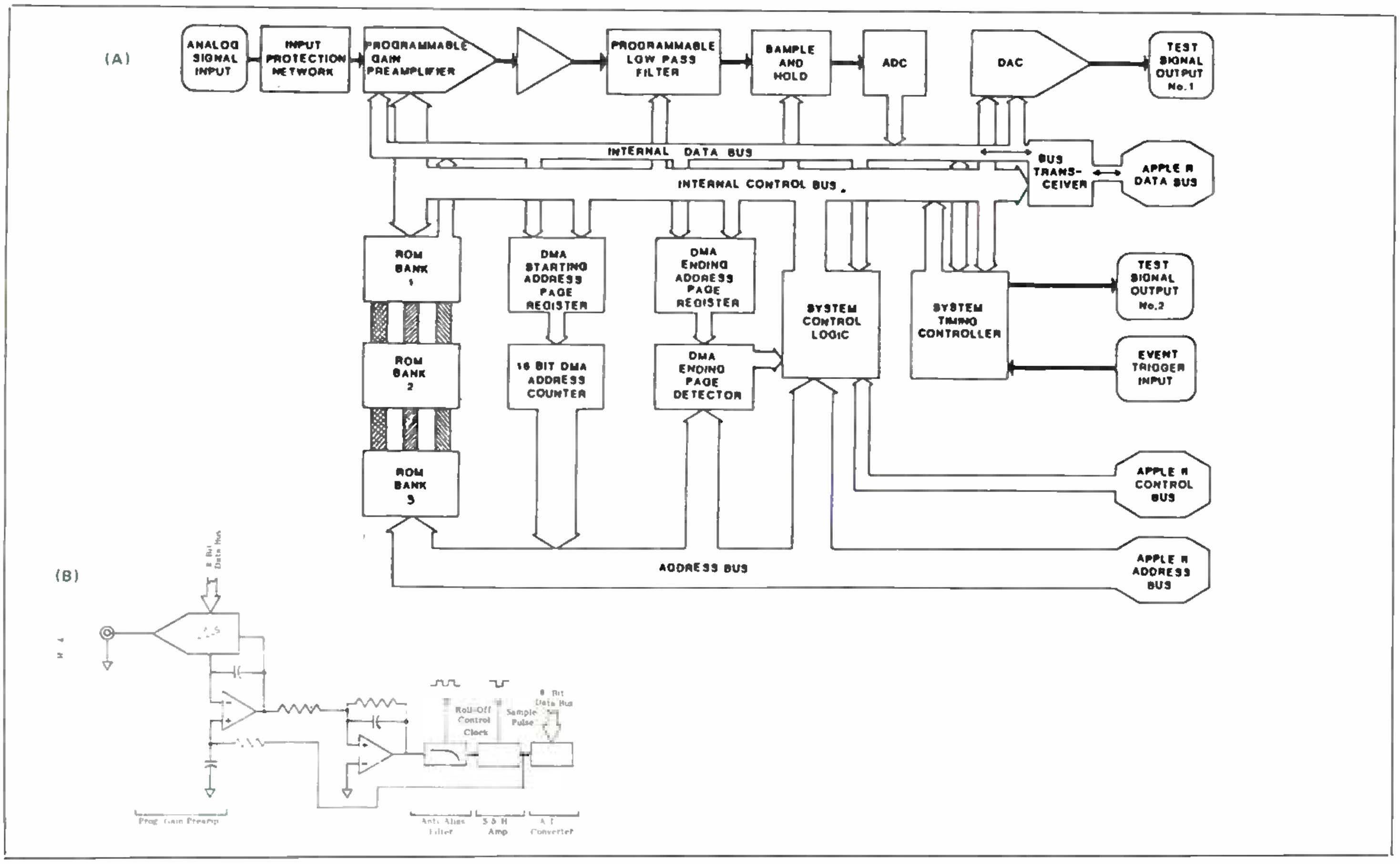


Figure 1. Block diagram of IQS Series 401 hardware (A) and Signal Acquisition Processor (B).

sampling rate is selected, the ANTI-ALIAS filter and test pulse signal width are set. All of the major commands are shown in FIGURE 3. Once a command has been invoked, it is either processed or prompts (asks) in plain English for the desired setting to be selected.

DATA ACQUISITION

Upon "booting" (loading the control program into the computer), some of the variables are in a "default" setting; i.e., they will be set to predetermined values unless you elect to change them. These are the A, C, S, and V commands. Pressing the space bar will initiate a test pulse and then take

SELECT SAMPLING RATES AND CORRESPONDING HF CUTOFF

(1)	60.0 kHz	30.0 kHz
(2)	46.5 kHz	23.24 kHz*
(3)	23.2 kHz	11.6 kHz
(4)	9.3 kHz	4.65 kHz
(5)	4.65 kHz	2.32 kHz
(6)	2.32 kHz	1.16 kHz
(7)	1.0 kHz	0.5 kHz
(8)	0.2 kHz	0.1 kHz

*Default

Figure 2. The "sampling rate menu" provides eight options from which to choose.

ACQUISITION MODE

CONTROL OPERATION

SPACE = SAMPLE data (4096 points)
E = ERASE screen
B = BLANKING of waveform (toggles)
A = AMPLIFICATION
C = CHANGE # of averages - in powers of 2
D = DELAY before sampling (1 = 0.6 msec)
P = PLOT screen to printer w/Grappler
S = SAMPLE RATE
V = VOLUME of test signal
I = INITIATE -collect & average data (2048 points)

ANALYSIS MODE

CONTROL OPERATION

RETURN = READY/DISPLAY FIRST 256 points of time data
1 = 128 point FFT
2 = 256 point FFT
3 = 512 point FFT
4 = 1024 point FFT
SPACE = RETURN to ACQUISITION MODE
D = DIFFERENCE - current less stored spectrum
B = PHASE and then GROUP DELAY
! = CONVOLVE - current with stored spectrum
Q = SPECTRUM DECAY PLOT from current waveform

DATA MANIPULATION

H = SHIFT data left
O = OUT DC - removes DC from time data
R = RESAMPLE
S = SKIP 256 points in time data
T = TRUNCATE time data
U = NEWSTART - select new start in data
W = WINDOW - cos wgt to first 256 points
Y = DIGITAL FILTER - moving avg. smooth
X = EXCHANGE spectrum in temp. storage
Y = DIGITAL FILTER - moving avg. smooth
% = SMOOTH spectrum

DISPLAY MANIPULATION

L = LOG display
P = PLOT screen to printer w/Grappler
RIGHT => = VIEW right half of 1024 point FFT
LEFT <= = VIEW left half of 1024 point FFT

Figure 3. System commands provided by the IQS 401-1 FFT Analyzer.

in one sampling of 4,096 points. While you are sampling data, the levels and time positioning of the analog input are displayed as a waveform on the monitor, just like an oscilloscope. If loudspeaker measurements are being taken, and there is a transit time of the test signal from the loudspeaker to the test mic, "D" may be pressed and the computer will prompt you with "DELAY IN MILLISECONDS?". The "return key" can be used at this time to display the first 256 points of time data on the monitor. However, if the environment is found to have insufficient signal-to-noise ratio, the "I" command can be used to eliminate the ambient noise by summing and averaging in order to increase the signal to noise ratio in excess of 20 dB. The technique of summing and averaging is well known (2). At this point, you may save the waveform to disk, or proceed with data processing.

DATA ANALYSIS

With a waveform in memory or one retrieved from disk files, FFTs can now be run. There are four different lengths of FFTs consisting of 128, 256, 512 or 1,024 points. The time length will be dependent upon the selected sampling rate and how many points in the FFT. Once an FFT has been computed and the power spectrum displayed, the phase response and group delay may also be computed and displayed. This is how simple it is: less than half a dozen keystrokes will give you all this information!

DATA MANIPULATION

If you want to use some signal conditioning, rather than using only the "raw" input data, DC offset can be removed from the signal. It can also be digitally filtered, resampled to improve the low-frequency resolution, the spectrum may be smoothed out, the signal may be windowed (weighted), etc. A very useful feature of microprocessed FFT is that any part of the captured waveform can be analyzed: the initial impulse (or early/direct sound), any subsequent reflection, or any combination of these!

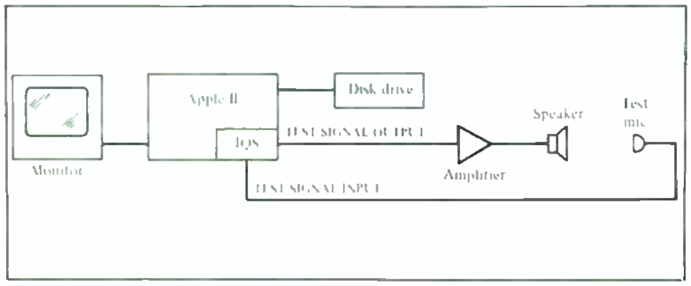


Figure 4. Instrumentation hookup.

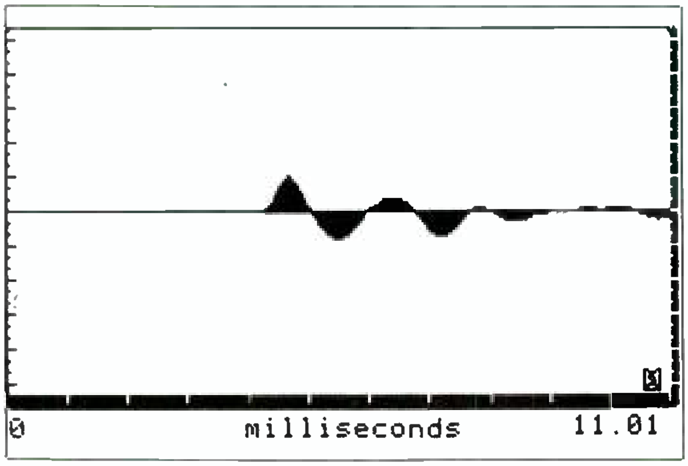


Figure 5. Time vs. amplitude information of woofer, where the vertical line represents volts in 1-volt increments.

Difference plots of stored data versus current data may be performed. This becomes an invaluable tool for before and after pictures, various comparisons between stored-on-disk information and device(s) under test data, etc. All FFT analysis, since it is a mathematical process, is performed in linear frequency scales. The addition of a movable cursor allows for an exact readout of both frequency and level. The advantage of using linear scales is that you can see the bandwidths of notches or peaks. And, if you'd like to see the display in log, simply press "L"!

At any point during a measurement session, any waveform or graphics display can be stored to disk for future use. Similarly, any waveform or display can be sent to a graphics printer or plotter. All of these commands are simple single keystrokes with prompts of NO or YES, the program prompts you with "TITLE?".

TEST SIGNALS

Any analog signal may be analyzed regardless of its waveshape (sine, cosine, square, etc.). It may be voice, low frequency vibration, the brain's alpha waves, musical instruments—any source! For loudspeaker testing there are several different types of test signals available, the most common being swept sine waves and pink noise. For the most part, swept sinewaves and pink noise will only tell part of the story; that is, they will show the steady-state condition of a system's transfer function.

Fortunately, music and natural sounds have more than just steady-state conditions. Primarily they are attack, steady-state, and decay, and they are characterized by pitch, duration, articulation, loudness, timbre, etc. Here is it interesting to point out the work of Risset and Wessel (3, 4). They found that if the attack portion of a sound is removed, it may become impossible for the ear to discern its origin!

Historically, transient test signals have been used for acoustic, transducer, and electrical analysis. Figwer (5), Cann and Lyon (6), Lincoln (7), Berman and Fincham (2), Peus (8), Hall (9), Fierstein (10), Lubin and Pearson (11), and Schaumberger (12, 13) are a few good examples of impulse testing applications.

If one would like to perform filtered and/or delayed swept sine wave measurements, one can use the external trigger function. Since the IQS FFT has an onboard test impulse signal regeneration circuit, the need for external test gear is eliminated. Referring to FIGURE 4, the simplicity of instrumentation hookup is clearly seen. However, and most importantly, the impulse response of the transfer function of a loudspeaker will immediately show its sound reproduction characteristics, i.e., transient, decay, resonances, ringing, even comb filtering! Since all musical sounds (even bowed hole notes of stringed instruments) have attack and transient portions of their signals, the impulse method will clearly display an excellent correlation to *the real world!*

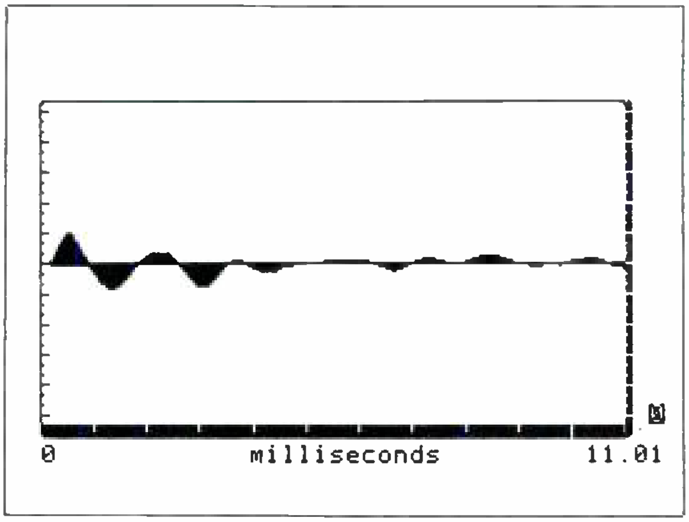


Figure 6. Repositioning of the data in Figure 5 by using the delay command.

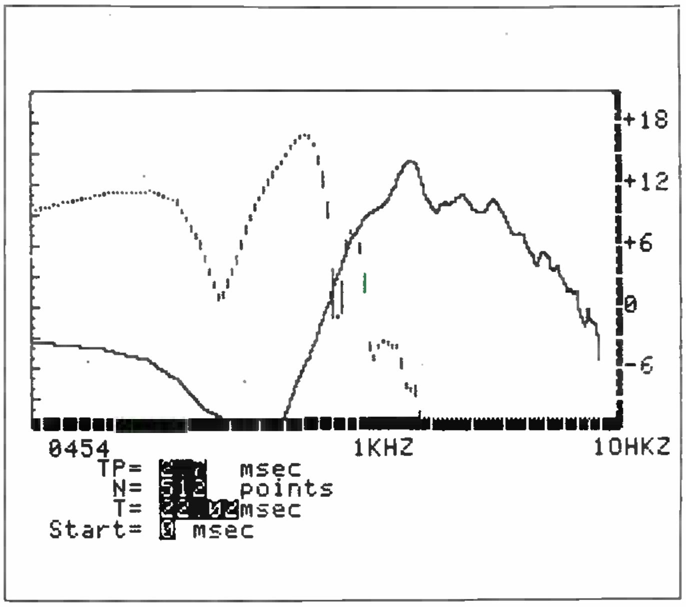


Figure 7. Simultaneous display of 512-point FFT analysis of two loudspeakers.

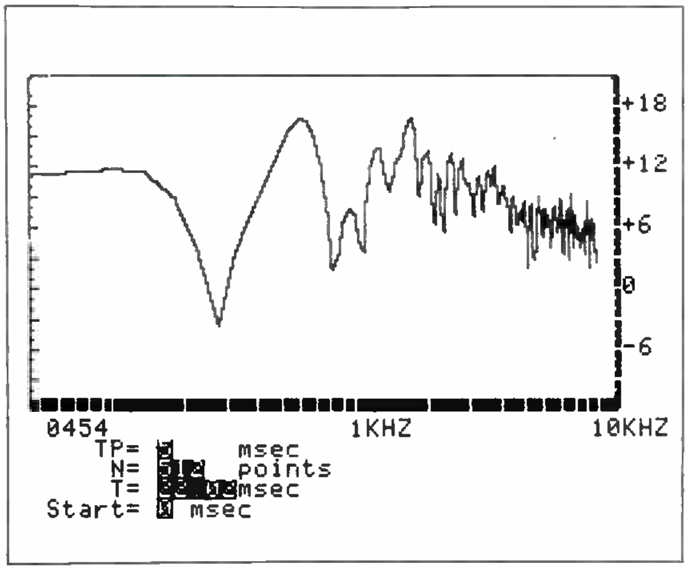


Figure 8. FFT analysis of the system with an 800 Hz crossover and a tilt EQ of 6 dB at the top end.

REAL WORLD APPLICATIONS

Before commencement of any measurement, the device(s) being tested must be clearly understood in terms of how they interact with their environment. Now that we have a basic understanding of the FFT analyzer, let's run through some practical applications. We'll measure a 15-in. woofer and a 1-in. throat-compression driver (on a 90 × 40 horn).

FIGURE 5 shows the first 11 msec. of time versus energy information of the woofer, as seen by the mic (including transit time). FIGURE 6 shows the repositioning of the same data by using the "D" or delay command. This data was gathered by summing and averaging 8 samples. This was more than adequate to overcome the ambient noise in my office. (Unfortunately, that's about 65 dB.) Once the data has been gathered, it may be stored to disk for later use.

Now we shall compute a 512-point FFT of both loudspeakers and display them simultaneously (as shown in FIGURE 7). FIGURE 8 is an FFT of the system with an 800 Hz crossover, and a "tilt" eq of +6 dB at the top end. At this point, phase response and group delay may be computed and

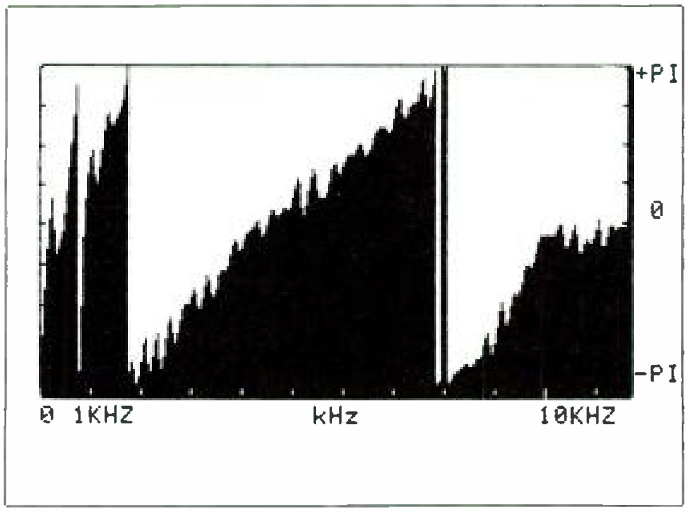


Figure 9. Computed phase response.

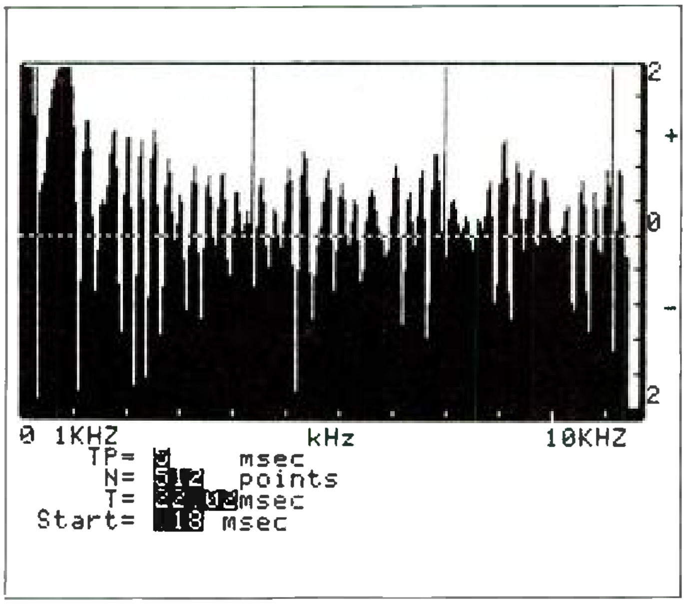


Figure 10. Computed group delay.

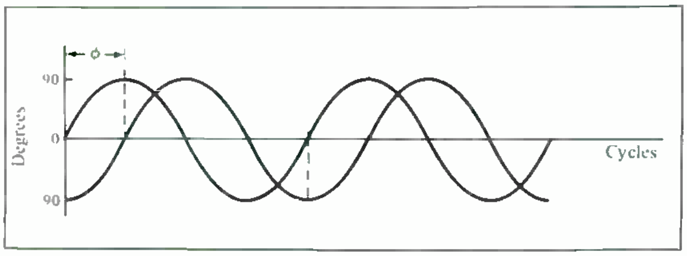


Figure 11. Example of phase shift, where 1 kHz sine waves are 90 degrees or .25 milliseconds apart.

displayed, as in FIGURES 9 and 10. Phase shift between the input and output of a device is related to the time of transmission, expressed as phase shift = time delay × 2π the frequency. Phase shift at a particular frequency may be directly expressed in time. Thus, as shown in FIGURE 11, a 90 degree phase shift of a 1000 Hz sine wave corresponds to a quarter of a cycle, or 0.00025 seconds.

Looking at the above equation, it can be seen that the phase shift corresponding to a given time delay increases with frequency. In order to avoid distortion, the time delay must be the same for all frequencies, or the curve of the phase shift plotted as a function of frequency on a linear scale must be a straight line increasing with frequency. Flat group delay or perfect time alignment are other ways of stating these functions.

SUMMARY

As we have seen, it *is* possible to narrow the information gap. The IQS FFT analyzer enables us to explore what is actually happening in real situations. Once the basic techniques of FFT analysis are mastered, the applications are endless. In this article, we have touched upon only a few.

A teacher of mine once told me, "I can tell you more about a sound system with a ruler than you can with all those fancy gadgets." In other words, we have to understand not only sound systems, but also the acoustic environments we are measuring. With tools like the FFT analyzer, and a good understanding of the techniques involved, we can indeed join the "scientific" and "subjective" worlds. But before we can do that, we must become familiar enough with our tools, so that they are not just fancy gadgets, but rather, practical rulers.

Editor's note: Next month, author Klapholz returns to these pages to put FFT theory into practice.

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Premiere Issue

PROFESSIONAL AUDIO:

The State of the Art

The *Sixteen Years Ago* Recording Studio

—George Alexandrovich*

This state-of-the-art picture of the recording studio is the preamble to a series under preparation. In this installment the author ranges wide to cover much of what is likely to be encountered in the studio. Future installments will tend to concentrate on specific areas of interest.

The past decade has seen advances in the field of electronics no one has expected. New discoveries in the field of semi-conductors have led to new technologies and changed most of the design concepts overnight. Gradually new technologies are making their way into the audio field, awakening audio engineers and specialists to the advantages of the new concepts over the older kind, a kind inferior in performance reliability and convenience.

The advent of semiconductors into audio closely coincided with the development of the stereo disc. This led to further developments of multi-track tape recording. Such tape machines in turn led to the redesign and rework of the mixing consoles and control rooms. As a result, the problems of the recording engineer and maintenance man suddenly increased two fold. Several channels of audio now had to be recorded simultaneously and monitored at the same time, each channel individually as well as the total mono mix, to be sure of proper phase relationships. Instead of caring for one channel only, maintenance had to be pulled on several, keeping proper balance, frequency response, phase and separation.

Disc cutting rooms had to be fitted with new equipment capable of stereo. Recording and cutting personnel had to be trained to use stereo equipment as well as to maintain it. Enormous new problems arose when semiconductor circuits were applied to the design of audio equipment, because of the inability of a majority of maintenance men to grasp immediately the basic operating principles of these new circuits. For quite some time a negative attitude existed among audio engineers and management toward the transistorized equipment. True, the first transistorized circuits were not designed with the same degree of sophistication as they are now, at times leaving more to be desired from their performance. At this time, the advantages of transistors over tubes have still not been recognized by many professionals. Many not yet set to accept semiconductors are those that were probing for faults in the equipment available at an earlier time and, as a result, continue criticizing it. Some of these criticisms are utterly ridiculous; for example, the claim that transistorized equipment does not produce "air around the sound."

I can prove that this is not so and that transistorized equipment can outperform tubes in distortion, frequency response, noise, in stable operation and reliability. As far as the air around the sound is concerned I need only say that transistorized circuits with their extended performance range reproduce information more faithfully than tubes without creating any side effects. True, they reveal more faults in the recording than do tubes, leaving the impression that tube circuits are cleaner sounding because of poorer transient response and restricted frequency range.

These innovations and changes in operating as well as in maintenance procedures have met strong opposition for another reason. Up to this time, recording was more often an art, with skill in it acquired more through experimentation and cut and try methods than through a proper scientific approach and knowledge of what conditions have to be met in order to achieve good recording. Today, new technologies require more science prerequisites for the audio man; he must have more theoretical knowledge and skill.

Advances and improvements have been made in every branch of audio recording. The most significant are in the design of the *equipment.* This inevitably has affected *operating procedures* and *techniques.* With new equipment and new technology *maintenance* gets the lion's share of the changes in procedures and requirements. It is my intention to use these three topics for future discussions, developing each one into an individual review of the tasks and problems facing the audio recording engineer today.

And those problems are numerous. As systems become more complex, operating procedures have to be more precisely controlled, maintenance of unfamiliar equipment and circuits becomes more painful. It is my goal to guide you men in the studio, behind the mixing consoles, and in front of the editing machines or disc-cutting lathes, to the correct approach used in solving the myriads of individual problems. I would like to shed light on the facts about the new equipment and operating methods showing how much their flexibility offers chances for a successful session. I want to cover the methods of preparation of the studio and control room for a session; how to conduct the session; proper storage of recorded information; mixing and editing; preparation of the master tape and master disc. Some topics will be discussed in greater detail than others; I hope to be able to share with the reader a few "trade secrets." These

**Vice-President, Engineering*
Fairchild Recording and Equipment Corp.
Long Island City, N.Y.

secrets are nothing more than short cuts to the solution of problems or ways in saving money that may turn out to be important to smaller studios with limited budgets. I will direct my efforts to be as down to earth in these discussions as possible so that every recording man can understand all that is being talked about. And there is a lot to talk about.

Today you can hardly find a studio not already in possession of a multi-track recording system (or at least contemplating acquiring one). A few studios are still operating on two tracks, more are set for four tracks and some for eight tracks. Since eight-track machines are possible because of miniaturization with transistorized equipment quite a number of professionals are now thinking of eight and even fifteen or sixteen tracks on one- or even half-inch tape.

The advantages of an entry into multi-track narrow tape recording should be obvious to every engineer as well as his management. The ability to record virtually every microphone on a separate track offers an easy remixing job with better chances for correction if during the take balance between the microphones was other than acceptable. Equalization, reverberation, and other effects can be added at will to an individual instrument or groups.

With the help of selective recording, better known as *selsync*, recording on the multi-track machines can be economical. One or more tracks can be recorded independently of each other. When the remaining tracks are to be recorded, the previously recorded ones are played back through the record head and fed into a separate circuit. This output is fed into the headphones placed on the heads of performers so that perfect synchronism is achieved between all tracks. In this way a few performers proficient on several different instruments can be used, eliminating the need for large group, yet achieving the same results. If the take is unsuccessful one track or any number of selected tracks can be erased and re-recorded again without affecting other tracks.

Almost all of the newer consoles incorporate separate equalization on each microphone channel, with separate echo or reverb feed, sometimes compression, as well as many other features. But it is important to know how and when to use these features. For instance, compression during the original take should be used *only* as an overload protection rather than for altering the dynamics. It means that the threshold of compression should be set above normal operating levels in the console. Equalization should also be used with discretion. In the original recording it should be used only to improve crosstalk by restricting the frequency range or as a means to better noise figures.

With multi-track recording there is a strong trend to record and store audio information with the least amount of deviation from the original sound. This way original performance is always at hand and special effects can easily be added during the remixing session.

The storage medium for original recording or tape has also seen numerous improvements. Electrical as well as physical properties have been affected. Tapes are manufactured today from better materials (Mylar) and coated with oxides capable of carrying higher magnetization forces and producing lower electrical noise when fully demagnetized. They also offer less friction with the recording heads. All of this produces a wider dynamic recording range.

Work is being done on high frequency bias to lower the hiss level normally generated by the bias currents. Networks, producing predistortion into the recording, compensate and cancel 3rd harmonic distortion caused by the tape itself when it is being recorded with levels approaching saturation of the oxide. Naturally this predistortion would vary with the type of tape used and should be adjusted for each individual brand. This technique can improve the dynamic range of the recorder up to 6 dB with attendant low distortion.

To ease the tasks of phase control, equipment has been developed to insure proper mike placement in the studio. Phase detection monitors are used in some installations to avoid phase cancellations at low frequencies.

All these improvements allow the recording engineer to work with wider margins of safety for better recordings.

A great deal has been accomplished in the past decade in the field of tape recording. Nevertheless, disc recording and disc pressing are still with us and are sure to remain for a long time to come. With the advent of the stereo disc, a multitude of new problems arose. Precision control of groove geometry and position as well as stylus alignment for best channel separation and minimum distortion calls for an increased ingenuity by the cutting engineer or technician. Earlier I talked about the conversion of an art into a science in general, but this part of sound recording belongs in a separate category, since there is still much "know-how" and ingenuity as well as experience required of a cutting man. Every step in setting up for cutting stereo is a critical one. In order to achieve optimum results in both the sound and the appearance of a record one must be thoroughly familiar with all facets of this skill. This part of recording contains the most trade secrets and shall be treated as such in a future separate section on disc recording. Many recording engineers can benefit from the information that has been assembled from many cutting rooms and as many ingenious operators and technicians.

I have reviewed hastily the basic problems of sound recording and some of many improvements that have been made in this field in the past few years. But no equipment is immune to mishandling or misuse. One of the biggest handicaps of studio setups today is the lack of maintenance and quality control. This might be through incompetent personnel or simply because of an absence of trained technicians.

It is quite common to find multimillion dollar installations without a good 'scope or signal generator; never mind looking for a distortion analyzer. Commonly, there is an absence of any maintenance records or studio block diagrams.

On numerous occasions it has been found that because of the lack of maintenance or negligence the best equipment was operating as poorly as the worst kind. Hum, noise, susceptibility to clicks, crosstalk, and intermodulation distortion have been found as results of wrong terminations and a lack of proper grounding. Correct phasing, proper wire identification, good soldering, and wire dressing as well as equipment location, are major items which shouldn't be forgotten. Maintenance may well make or break the studio, so I will continually place special emphasis on proper maintenance.

Let this short review be an introduction into the coming series of talks about the wide field of practical professional audio engineering.

GEORGE ALEXANDROVICH

Sound Recording and Sound Processing Technology—Sixteen Years Later

Author Alexandrovich has returned to offer us a look at what has transpired (audio-wise) in the intervening sixteen years.

IN THE PREMIER ISSUE of db Magazine, exactly sixteen years ago, I gave a general overview of the state of sound recording and sound processing technology. At that time, tube-type equipment was becoming obsolete and everything was being converted to solid-state circuitry. Integration as we know it today did not exist. All circuits were built around discrete semiconductor devices. Though we were busy converting everything to solid-state devices, we were not in a position to utilize all of the advantages transistors offered us.

Today, almost all of the elements can be synthesized and represented by integrated circuits. This means that there is no need to wind huge chokes for our filters, no need for super large electrolytics, high power switches, or large output tubes. We can have all this in miniaturized form. Integrated circuits or discrete devices now have performance specifications that put to shame even the most advanced devices of yesteryear. We can drive high-voltage circuits from single-cell batteries and convert any type of power source to any other type we wish. Mechanical meters are being superseded by LEDs or LCDs.

SWITCHING DEVICES

Switching can be accomplished by special high-speed transistors, FETs (Field Effect Transistors), triacs, diacs, silicon-controlled rectifiers, or a variety of other modern switching devices. For instance, the technology of power switching has reached the point where power companies have started using solid-state MOSFETs to switch currents from high-voltage lines to low-voltage lines. The same devices are being used to operate brushless motors, where the current in the field winding, rather than being controlled by the collector, is being controlled by a combination of hall effects devices and solid-state switches. These can be operated remotely with a minimum amount of power, yet with incredible efficiency and reliability. The switching speed of these devices is also incredible. Hall effects devices are now used to sense magnetic fields, performing functions magnetic tape heads did before. And the work to perfect these devices is still going on.

LIGHT SENSORS

Old-fashioned bulbs and pilot lights have been replaced by almost indestructible LEDs, which are available in several colors and a variety of shapes and intensities. We should also mention the wide choice of light-sensing devices available today. There are light-sensitive transistors, diodes (opto type), SCRs, FETs and other sensors such as cadmium sulfide and selenide cells.

Naturally, we should not forget about lasers as a source of light. An entirely new branch of audio recording and playback is based on the laser beam principle. (More about that later.) There are high- and low-power lasers. High-power lasers can be used to cut steel plates, knock missiles out of the sky, or as a cutting or cauterizing tool. Laser beams can weld tissue, remove cataracts, cut away malignant tissue and free clogged blood vessels. And who said that the laser is dangerous to human health?

Closer to the subject at hand, laser-based fiber optics are now being used to carry large numbers of multiplexed channels of information and audio signals.

AMPLIFIERS

Micro-miniaturization in all aspects of technology has revolutionized amplifiers. Operational amplifiers (or op amps) are everywhere, with performance we could hardly have dreamed about some 30 years ago. As long as we know what we want and what kind of performance we require, we

George Alexandrovich is vice president, Field Engineering and Professional Products manager for Stanton Magnetics Inc. He is also on the Board of Governors of the AES.

can buy it (maybe not right away, but eventually). It seems that the demand is so great for all types of devices these days that semiconductor manufacturers are quoting delivery dates of six months to one year.

Introduction of VSLI (Very Large Scale Integration) technology has had an enormous impact on designers. Coffee makers, TV games, computers, calculators and cash registers are all being converted to chips with either ROM or EPROM memories (read only and read and write types). Audio chips have appeared that contain entire noise-reduction systems, equalizer and filter circuits, control logic for drives and switching, as well as automation systems for remixing music from multi-track tape recorders.

Awash in this sea of new devices is the one that is arguably the most valuable device for the sound-mixing engineer: the VCA (Voltage Controlled Amplifier). This replaces the conventional gain control fader, increasing the reliability of the gain control function while providing the ability to control gain from other channels and gang as many channels as one desires. It can also function as a perfect switch or gradual automatic fader.

THE CONSOLE AND THE MIX

The size of all these devices allows us to pack an entire multi-channel console into a space the size of a lunch box (excluding the knobs and manual controls). The size of the controls is still a problem. Although we can design the console to be fully automatic, it can never duplicate the decision-making ability of a recording engineer. The engineer still needs the necessary interface between his/her hands and the electronics; fader knobs have to be there to tell the engineer about the setting of the mic gain or sub-channel status; switches must have indicators to tell him what is connected where. Even with all these technological advances the human factor remains (thankfully) of prime importance. The laws of physics and acoustics have not changed; nor has the desire for good music.

The right choice and balance of sounds still depends on the taste and skill of the mixing engineer and the producer who have to know how to utilize all of the new marvelous tools technology has presented us with. They have to be artists, sensitive to the desires of the public, and yet creative, innovative and inventive. The trend today in recording seems to be toward having as many input channels as there are instruments and microphones on the floor so that sound can be recorded indiscriminately at the highest possible level without distortion for best signal-to-noise ratio.

After the recording session ends, the real work begins: the remixing. With the increasing complexity of mixing consoles, it is no longer a matter of mixing signals from two strategically placed microphones from a single-take session. It is a declaration of war on knobs and controls—a chess game with the console with no known winner. The boards are so complex today that each module is like a miniature console with equalizers, post- and pre-echo sends, solo circuits, assignment buses, or microprocessor controls.

Finally, after weeks of adjusting track levels, numerous variations of equalization, sweetening, flanging effects or reverberation with delays, we are ready for the last step: final mix to two channels. And what is the end product of all this? An LP disc or a cassette. In essence, then, what it all boils down to is that this multitude of channels ends up (hopefully) as a high-quality two-channel program.

TAPE RECORDING TECHNOLOGY

While improvements in technology were taking place, the quality of tape recordings was also advancing. Not only were new and better tape heads designed and built, but the tape itself was greatly improved. New types of oxide and magnetic coatings on plastic films have allowed the quality of sound from cassettes to overshadow and eliminate the competition: the 8-track cartridge and open reel tape. The advent of Walkman-type cassette players has captured young audiences and created new markets for high-fidelity sound.

THE LP VS. TAPE

The LP has not enjoyed the same success—but not because of a lack of technology or hardware. Rather it is a result of the overconfidence of record producers who were so sure that the record was beyond competition. The disc was considered to be such a perfect medium that it was felt that no matter how much we abused it, it would hide our mistakes. Not so. No one in the tape recording field would drive the tape recorder into overload. On the disc however, there seems to be no limit for the modulation level. When recording on tape, the best tapes are generally used. In disc pressing plants, though, cost is the main factor. Thus, the cheaper the material, the happier the record manufacturers.

LASER TECHNOLOGY

The technology of digital recording grew as a result of this disregard for the quality of the analog disc. In using the laser beam, the recording was achieved on a 12 cm disc by converting the audio signals into digital form so that from the moment the A/D conversion took place until the playback circuits converted the stored information back to analog form, no signal deterioration took place. Just to overcome the surface noise of the inferior polystyrene records (all 45s are made this way today) and reduce distortion and mistracking created by the severe overmodulation of the groove, engineers have developed a new system using binary numbers consisting of 1s and 0s to represent analog signals. The signal is chopped up into 44,100 parts per second, with each part converted to a binary number of 14 digits. $14 \times 44{,}100 = 617{,}400$ bits of information used to represent audio waveforms during the 1sec interval. Since the sampling at higher frequencies remains the same, a 10 kHz signal, for instance, is sensed only 4.1 times during its cycle. In converting the digital information back to analog, the high-frequency information will only be an approximation of the original 10 kHz waveform. Nevertheless, the system works.

In order to store this enormous amount of data on the 12 cm disc, the linear speed has to be very high—forcing the disc to rotate as fast as 500 revolutions per minute. The width of the track produced on the flat digital compact disc is less than 0.5 microns wide, with constant spacing between turns of 1.6 microns. The digital information is burned into the surface of the master by means of a laser that is pulsed to produce pits for the 1s and leave the smooth surface for the 0s.

There are several advantages to digital disc and digital sound recording technology, including lack of signal degradation, low distortion, excellent signal-to-noise ratio, wide dynamic range and separation between channels. There is also a very sharp, abrupt loss of all frequencies above 20 kHz. And, because the frequencies are synthesized, one has full control of how to re-create them. There are numerous circuits built into the digital equipment to correct for loss of bits of data and to eliminate the harmful effects of sampling frequencies and filters used to control the purity of the sound. Along with audio signals, all sorts of additional information is encoded, e.g., signals to control the speed with which the data is stored and then released, as well as information to control channel separation and levels. The cost of such a system is great, and the disc manufacturing process is precise and complex.

WHY CAN'T WE DO THAT?

If we compare the cost of manufacturing analog LPs with that of the digital compact discs, the LP wins hands down. A high-quality analog disc costs no more than half a dollar—minus jacket and royalties in small quantities (1000 pcs)—while the digital disc may cost 10 times as much. And this cost will not diminish greatly in the future.

There is one thing that I fail to understand: The auto industry has turned down exotic engines such as turbines, steam engines, electric, and other types of power plants for good ol' reliable internal combustion engines. They took old gas guzzlers and, via computer control and careful engineering, turned them into modern engines that pass all economy standards and pollution-control requirements. Why can't we take all the technology we possess, including the computer and digital know-how, and glorify our good ol' reliable analog disc?

Couldn't we use good materials that would repel dust instead of attracting it? Couldn't we combine noise reduction ideas like Dolby, dbx, CX, DNR and others to achieve the same results we do with the CD? Why can't we control the speed of the turntable with supersonic frequencies to change the speed of the motor, while using error- and distortion-correction circuits that we already know how to construct and use?

The simplicity of analog record manufacturing is astonishing. In addition, the investment is small. And it is this last fact that may hold the key to the future of our industry. I believe that the analog record is here to stay—if we put our minds to it. The improvements needed in the quality of the disc are relatively straightforward: better plating care and more electronics to mask harmful effects. Isn't this what is done for a digital disc?

A positive sign that this improvement of the LP is beginning to happen is the development of the DDM (Direct Metal Mastering) process. This new technology is a refinement of the old method of mastering the analog disc by cutting the metal. (Edison was embossing the metal using a tangential tracking tonearm in his first machine 100 years ago, and played it using a doorknob stylus which today we call elliptical.) If technology keeps coming back to the same methods, we must have been doing something right all these years.

I cannot help but recall a session at the recently concluded AES Convention in New York City. A/B tests were conducted between digital and analog records in front of an audience of seasoned professionals and distinguished recording engineers. The conclusion that was reached after the meeting showed that the majority of the participants could not distinguish between the two sources of music. This conclusion would seem to indicate that the consumer is going to need some very good reasons to justify the cost of a digital system before he/she would buy it.

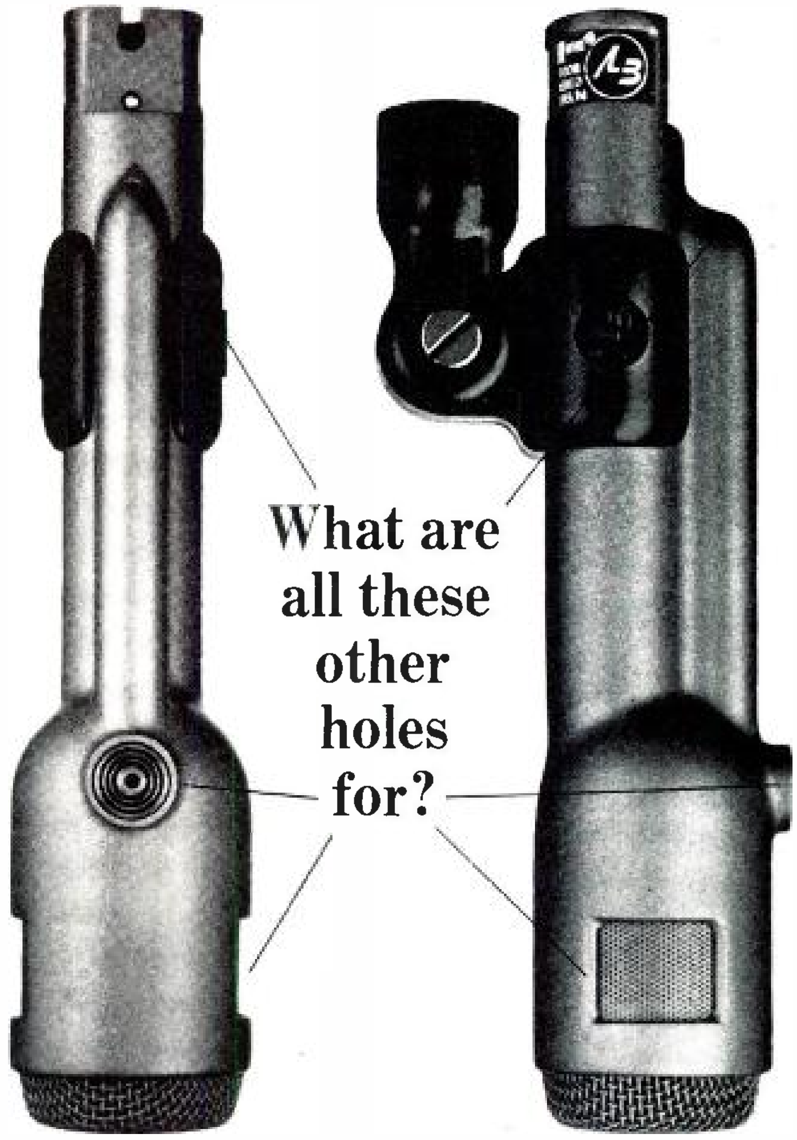
But then, what have we done for the consumer in the past 16 years? He listens to headsets and a Walkman when he jogs, uses the same stylus for his records that he used a decade ago (he forgot to change it), probably bought new speakers because he liked the walnut veneer—and still doesn't know if they are correctly placed in his living room. He heard the word digital and associated it with the label on the analog LP. Now, when the "real thing" comes along, he will be more than puzzled. He will be hearing the same selections he now owns in analog form, with much higher dynamic range and lower noise it's true, but he will have to turn the gain controls down because his neighbors might become very annoyed.

SOME HARD QUESTIONS

What I've said is not new. We have been aware of it for some time, but have not really thought about it. Well, maybe now is the time. Given all the facts, are we going in the right direction? Are we thinking logically and being practical in our deductions? Are we making everyone's lives more pleasant (which is, after all, what music's all about)? Or are we taking our technology further and further away from the consumer's reach so that someday the only type of entertainment the average consumer may be able to afford will be computer-synthesized music stored on a memory chip played through an electrode attached to the nervous system?

Think about it. ■

If the Electro-Voice Model 666 picks up sound here...



Sixteen Years Ago

The holes in the top, sides and rear of the Electro-Voice Model 666 make it the finest dynamic cardioid microphone you can buy. These holes reduce sound pickup at the sides, and practically cancel sound arriving from the rear. Only an Electro-Voice Variable-D® microphone has them.

Behind the slots on each side is a tiny acoustic "window" that leads directly to the back of the 666 Acoustalloy® diaphragm. The route is short, small, and designed to let only highs get through. The path is so arranged that when highs from the back of the 666 arrive, they are cut in loudness by almost 20 db. Highs arriving from the front aren't affected. Why two "windows"? So that sound rejection is uniform and symmetrical regardless of microphone placement.

The hole on top is for the mid-range. It works the same, but with a longer path and added filters to affect only the mid-frequencies. And near the rear is another hole for the lows, with an even longer path and more filtering that delays only the bass sounds, again providing almost 20 db of cancellation of sounds arriving from the rear. This "three-way" system of ports insures that the cancellation of sound from the back is just as uniform as the pickup of sound from the front—without any loss of sensitivity. The result is uniform cardioid effectiveness at every frequency for outstanding noise and feedback control.

Most other cardioid-type microphones have a single cancellation port for all frequencies. At best, this is a compromise, and indeed, many of these "single-hole" cardioids are actually omnidirectional at one frequency or another!

In addition to high sensitivity to shock and wind noises, single-port cardioid microphones also suffer from proximity effect. As you get ultra-close, bass response rises. There's nothing you can do about this varying bass response — except use a Variable-D microphone with multi-port design* that eliminates this problem completely.

Because it works better, the E-V 666 Dynamic Cardioid is one of the most popular directional microphones on the market. Internal taps offer 50, 150, or 250 ohm impedance output. Frequency range is peak-free from 30 to 16,000 Hz (cps). Output is—58db.

To learn more about Variable-D microphones, write for our free booklet, "The Directional Microphone Story." Then see and try the E-V 666 at your nearby Electro-Voice professional microphone headquarters. Just $255.00 in non-reflecting gray, complete with clamp-on stand mount. Or try the similar Model 665. Response from 50 to 14,000 Hz (cps), $150.00 (list prices less normal trade discounts).

*Pat. No. 3,115,207

ELECTRO-VOICE, INC., Dept. 1071 BD:
686 Cecil Street, Buchanan, Michigan 49107

Electro-Voice®
SETTING NEW STANDARDS IN SOUND

Sound Reinforcement Sixteen Years Ago

—Martin Dickstein*

The elements that make up a sound reinforcement system have come a long way since the days of the cupped hand. But the present state of the art is merely a stopping point. Where do we go from here?

Back in the very, very old days, it was necessary to be within "earshot" of a sound's originating point. Earshot depended for the most part on the frequency characteristics, loudness, wind direction, surrounding noises, terrain or location of source and listener, just as it does today, whether the receiving device is an ear or mechanism. In those days, though, earshot distance actually meant to an ear. There was no way to preserve the sound for later listening, nor was there any way to extend the distance. If anyone wanted to hear the sound they had to be there, fairly close to the source. Attempts to increase the "throw" distance through the use of cupped hands or other horn-like devices helped the situation a little. In time, it was learned that sounds could be directed somewhat toward the listeners if the source was located in front of something such as a hill or structure. Carefully shaped and strategically placed vases in the proximity of performers, prevailing winds, and higher but closer seating areas were used in further attempts to permit more listeners to hear the sounds originating from the source.

Through the many years that followed, more attempts were made to enlarge the listening area, but meeting halls, auditoria, concert halls, churches and ampitheaters had to be built to take advantage of natural acoustics to "spread the word." It was not until the availability of electronic amplification that something was done to substantially enlarge the listening area.

Recordings could now be made using a microphone which changed sound impulses to electrical impulses. These recordings could be played through speakers which distributed sound to listeners in a fairly good size area. When movies started to talk, though, a real need for "big sound" became apparent. Theaters could seat pretty big crowds to see the screens, but now it became necessary that all these people also be able to hear the voices, sounds, and music. Larger speakers, higher-output microphones, and more powerful amplifiers had to be developed. And they were.

This solved some problems but, as seems to happen in almost every field, created new ones.

As microphones came into use for addressing live audiences, methods had to be found to make the mikes smaller in order to avoid hiding the speaker or performer. Lapel microphones, lavaliers to hang around the neck, and button-hold mikes were designed to permit the speaker or singer free use of the hands during a performance. As sensitivity of microphones improved it became necessary to manufacture units with different pickup patterns so that some had very narrow cones of sensitivity, some could pick up through greater angles and some were made to be non-restrictive, or omnidirectional.

Many types of microphones are now made for music, offering better frequency response in the bass range or high-frequency range or with smooth response through a very wide band of frequencies. Some distort sound less than others during loud, impact sounds. Others have pre-emphasis in the voice range built in. Still others are made to pick up through an extremely narrow angle but at a greater distance. Each microphone manufacturer today tries to cover a particular market or use with a variety of types and shapes especially designed and engineered for each application.

For locations where microphones are used near loudspeakers or in highly reverberant areas, the choice is a combination of narrow pickup angle, smooth frequency response and high front-to-rear rejection ratio to diminish the possibility of acoustic feedback. Today, microphones are made with these singular characteristics, combining those qualities that make them particularly suited for individual requirements, in almost all sizes and shapes and at almost any price.

Amplification

Many changes in the construction of microphones were caused by, and resulted in, changes made in the amplifier units developed to raise the minute power output of the microphone to the level necessary to drive a loudspeaker strongly enough to enlarge the listening area by a substantial amount. Early amplifier designs for power applications were made to match existing microphones and speakers. The desired power output was achieved first with existing tubes and components through the use of some known circuits, then with modified circuits and newly developed tubes, new ideas on impedance matching, energy transfer, interstage coupling and feedback.

Inherent tube and circuitry noise was slowly eliminated with the development of new materials for use in filaments, grids, resistors, and the resulting modifications in power supplies and filtering methods. Frequency response was broadened to provide smoother lows and more natural highs. Components were designed to withstand the heat developed by high-power tubes.

Mixing circuits were devised to offer greater possibilities for multi-source originations. Special filters were designed to compensate for recording characteristics. Mixer units were separated from the power chassis to permit the more delicate low-voltage circuitry to operate to its own best advantage without interference from tube heat and power transformers. Bridging circuitry and matching networks were calculated and engineered to permit flexibility of inputs. Power amplifiers could not be located at a distance from the mixing unit whenever desired without any appreciable loss of quality or energy.

Amplifiers were modified to reduce distortion, increase frequency response, provide higher power and take up less space. Preamplifier mixers were also modified to operate with a greater frequency range and lower noise and distortion level.

As power ratings of amplifiers were increased to provide greater audience coverage, constant-voltage output systems were devised to allow more and more speakers to be added without loss of level.

Separate mixer/amplifier units and packaged, or combined, mixer-amplifier equipments are now available from many manufacturers in various price ranges depending on the application.

With the arrival of stereophonic reproduction, mixers and amplifiers have been again redesigned to permit two and three channels to be available on units built to the same dimensions as previous mono equipment, and with a minimum of cross-channel interference.

The disassociation of mixer from amplifier has permitted the development of equipment specifically to solve some of the problems incurred in multi-input and high-level speaker output systems. Limiters and compressors have come into fairly extensive use to prevent sharp bursts of sound from distorting the quality of reproduction or damaging the equipment. In the same way, expanders to raise the level of the source sound automatically if it should fall below a predetermined point, are now common.

Most recently, equalizers of various types have been engineered to permit a greater opportunity to emphasize desired frequencies and eliminate undesired ones. In reverberant auditoria greater intelligibility can be realized by raising the relative importance of certain frequencies and diminishing the level of others, thus limiting the discrete ones causing the disturbing reverberation and, as a result, permitting higher levels of output in the desired ranges. Less sophisticated equalizers are available that operate on calculated curves at fewer, but carefully selected, frequencies from those units that are specifically designed to permit exceptionally sharp operation at a great number of frequencies.

In recent studies of acoustics and reverberation, it has been found that if output frequencies can be made slightly different from the original ones, by an amount insufficient to be heard, then feedback can be reduced. This will allow higher output levels. The result is the development of a frequency shifter which raises each input frequency by 5 Hertz at the output. Levels can then be increased as much as 3 dB, or more, for greater audience coverage.

Solid State

In most recent years, preamplifiers, amplifiers, compressors and even power supplies have undergone a most revolutionary change with development of the transistor. In its infancy, the transistor introduced its own sound characteristic when used in audio equipment. It also could not withstand the punishment that a tube could when something happened in the internal circuitry of the equipment or in the speaker lines. With further development, this device is now being used in almost all equipment. New circuitry has been devised to keep the transistor from breaking down in the event the equipment becomes defective, or if trouble develops in the output lines, or even if there is a sudden power surge. Transistors of various designs and internal structures have been developed for special applications and while some are still being made that cannot be handled without special precautions, for the most part, these tiny units have made such inroads that they have almost taken over completely.

Audio equipment can now be made smaller, lighter, with very good frequency and noise characteristics, but not yet quite at the same price as the tube equipment.

Since the heat developed by a transistor is much less than that given off by a tube of equivalent operation, more equipment can be mounted in smaller housings. Nevertheless, all transistor equipment should still be protected from being overheated by any tube units which may be used in the same system.

Speakers

As other units of a sound distribution system have changed, so did the speaker. Extremely bulky, inefficient loudspeakers with coil-generated fields changed to smaller, lighter, more efficient units with permanent magnets. Cone material changed along with the type of suspension of open ends. Frequency response improved, rattle was eliminated in better speakers during peaks, shapes of magnets changed as new materials were found to create stronger fields in a smaller space, and efficiency improved as less power was required to move the cone. Speakers, as with microphones, have been developed for different applications, with unique specifications of frequency response and smoothness, power ratings, angle of dispersion, efficiency and size. Smoothness of the response curve is desired to reduce feedback when used in the vicinity of open microphones. Concepts of speaker housings and their internal structure, front openings and sizes, dimensions and material have undergone study and modification to improve particular characteristics deemed necessary by the variety of applications.

A study was also made of the interaction of sound waves when speakers were located close together. It was found that a mounting of several speakers in a particular arrangement provides characteristics unique to that arrangement. A linear array, with speaker centers a definite distance apart, creates a "spotlight" effect with the vertical angle less than that of a single speaker alone. That angle decreases as the number of speakers increases.

Sound columns are now used in areas where reverberation time is high to direct the sound towards the audience and to decrease the wasted sound energy which would otherwise hit the ceiling or floor and create reflected, disturbing sound patterns. Straight line and curved speaker arrays, multiple speaker arrangements, filter networks and other concepts have sprung from the original theory in an effort to produce units that will suit an application range or fit a price category. Still, specially designed units are required when the circumstances are sufficiently unique to warrant this engineering, as for example tremendous audiences in large outdoor areas or music halls where highest fidelity is required at each seat.

As more and more equipment has been developed and produced in each phase of sound distribution systems, it has become obvious that standardization is required. Microphone output impedance must match preamplifier inputs, amplifier outputs must match speakers, preamplifier outputs and amplifier inputs, as well as those of intermediate equipment, must also match for proper operation of a complete system. Characteristics of equipment, depending on several variables, have to be defined at particular points of measurement.

Prognostications?

Now that all these concepts, systems, equipment and many of the standards have become accepted, tried and developed to meet a great variety of uses and conditions, have we gone as far as we can in sound reinforcement and public address? Can anything more be done to improve what is now available to compare with the strides in development up to now? Further miniaturization seems to be part of the answer.

Will mikes be made with mini-transmitters to eliminate the use of connecting cables altogether? Will receiver-amplifiers be further diminshed in size through the use of multi-function modules so that they can be located at each speaker instead of in central racks? Will speakers become smaller to permit them to be hidden entirely? Will new studies of acoustics, reverberation, sound transmission and dispersion create havoc with present standards? Or will future studies of the ear and its operation, the response of the brain and greater use of psychedelic sound provide the new incentives for the developments yet to come? It might prove interesting to live long enough to hear for ourselves.

**Television Utilities Corp*
Long Island City, N.Y.

JOHN EARGLE

Sound Reinforcement—The State of the Art Sixteen Years Later

The following article (briefly) sums up some of the changes that have taken place in the field of sound reinforcement over the past 16 years.

IN THE SIXTEEN years that have elapsed since the first issue of db Magazine, the technology and practice of sound reinforcement has made some pretty large strides. The major impact has undoubtedly been the requirements of large-scale music reinforcement—a facet of the art barely known in 1967.

In the mid-sixties, professional sound reinforcement was dominated by a few companies, and their hardware was essentially that which had been developed some twenty years prior for the motion picture theater.

But while the hardware side of the business was relatively static, there was a good deal going on in the areas of system analysis and equalization. Paul Boner had established the criteria for system gain calculations, and his pioneering work in system equalization is known to just about everyone.

As the seventies progressed, we saw the rise of high-quality, high-level reinforcement of music. The rock concert had become big business, and its particular requirements spurred rapid development of new hardware specifically tailored for use on the road. Let us look at some of these requirements.

AMPLIFIERS

Amplifiers of the mid-sixties were made for fixed installations. They were heavy, did not travel well, and were modestly powered. While all of this was fine for most speech reinforcement applications, music reinforcement required new designs. Specifically, high power was needed, and Crown's DC-300 inaugurated the new era of solid-state high-power design. Still other manufacturers developed lighter weight

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John Eargle of JBL, Inc. and JME Associates, is a regular db columnist.

amplifiers using switching power supplies. The industry is still waiting for the definitive Class-D, or switching output, amplifier design.

INPUT EQUIPMENT

The speech input consoles that were used in the mid-sixties were little more than elaborations on broadcast designs. Today's music reinforcement consoles resemble those found in recording studios—a result of the expectation at the rock concert that the performers sound substantially like their records do! This means that all the tricks of signal processing that have become standard in the studio are now to be found in connection with large rock concerts.

MICROPHONES

Microphones have gotten better and cheaper during the last sixteen years. The electret design, in particular, has become quite popular, since it offers the benefits of the capacitor microphone at a relatively low cost. For sound pick-up in the theater or for classical music reinforcement, the various boundary-type microphones have become quite common. For vocal pick-up in the musical theater, vastly improved wireless microphones are now standard.

LOUDSPEAKERS

Just as amplifiers of the mid-sixties were power-limited for many of the applications of music reinforcement, so too were most of the low-frequency (LF) and high-frequency (HF) drivers of that day. After all, we must remember that these transducers were never meant to see more than 50 or 60 watts.

The first LF transducers to be pressed into service for music reinforcement were the various models that had been developed for musical instrument use, namely for electric guitars. These devices were highly colored—on purpose—and were capable of taking considerable abuse. What was really needed for high-level music reinforcement was a new series of products that combined low coloration with high power handling. Thus began an industry-wide quest for new high-temperature adhesives and better magnet structures.

The problems encountered with HF compression drivers were harder to cope with. Given new adhesives and surround treatments, they could take considerable power. However, distortion in HF drivers is largely a function of certain thermodynamic conditions, and this is something that cannot be corrected through attention to materials and mechanical integrity. It can only be solved through dividing the spectrum differently. Today, we see large-scale systems making use of special mid-range (MF) elements that have been designed specifically for the task of handling the 200 to 2 kHz decade.

FLAT POWER RESPONSE

Perhaps the most important conceptual notion of recent years has been that of flat power response. This term has been around only as long as constant coverage horns have been available. In essence, these new horns make it possible for the axial response of a system to be flat, while at the same time the total radiated power can be held nearly flat. This concept has also favored, for many applications, simple, ported LF systems instead of the large LF horns of earlier years. The benefit of flat power response is a smoother, less colored sound, and one that generally requires considerably less equalization than earlier systems.

BANDWIDTH EXTENSION

In the mid-sixties, the bandwidth of a music reinforcement system probably covered the range from 40 Hz to 10 or 12 kHz. Today, subwoofers are common, and their response, even in large outdoor environments, can extend down to the 25 Hz range. At the high end, ring radiators, used in multiples, extend the response out to 18 kHz.

WHAT OF THE FUTURE?

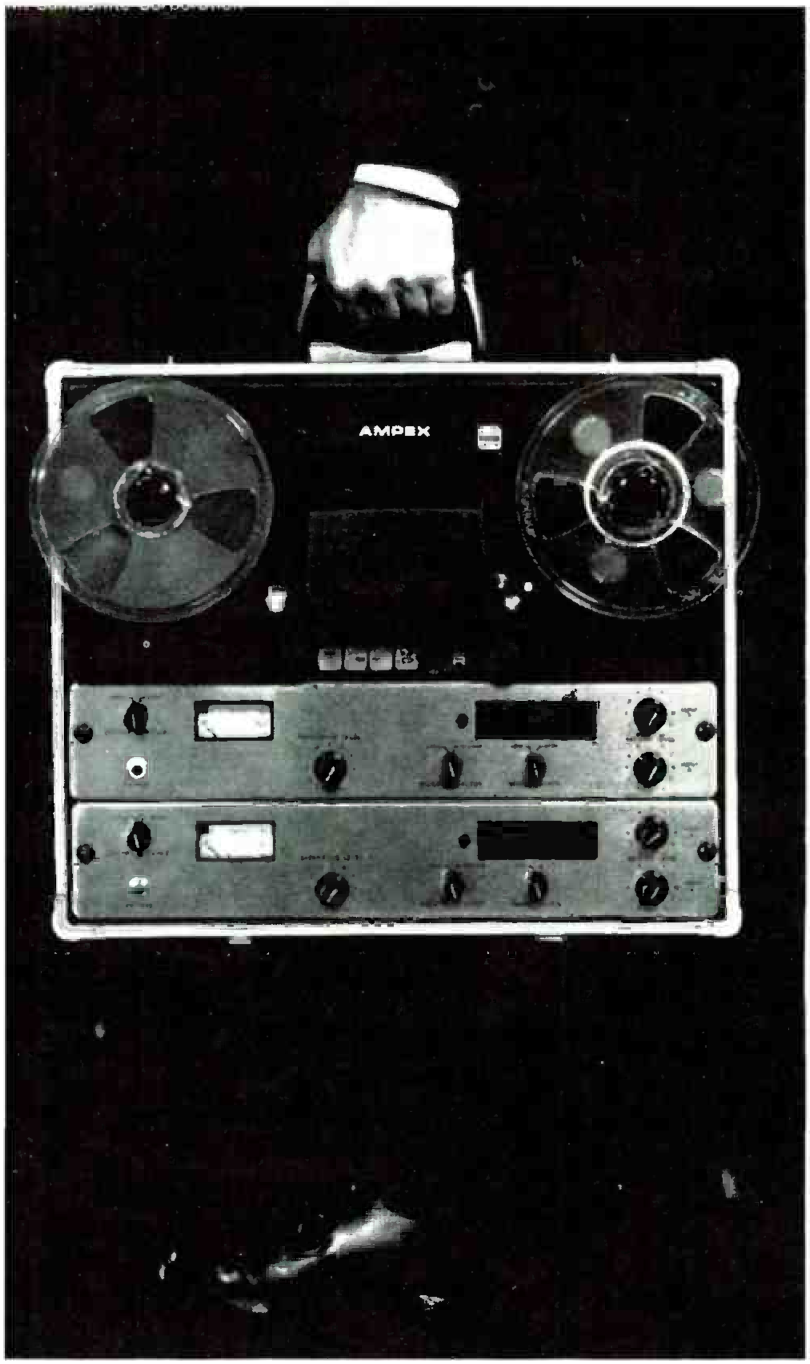
Most of the principles of transduction are pretty much in place, and have been for the last fifty years. We will undoubtedly see greater attention focused on lowering distortion, and it is quite probable that direct radiator devices will come into their own for all parts of the frequency range, except the top 1⅓ octave.

A great deal of the development will have a lot to do with how music itself evolves. Present trends emphasize musical detail and inner structure. Clean vocal lines are important as is a greater attention to the value of dynamic range in musical structure.

Some people feel that the large-scale rock concert will give way to smaller forms in smaller venues. If this is to be, then we will find greater attention to subtleties in music reinforcement, and there will probably be a good bit of attention given to multi-channel sound.

Lest we give the impression that the demands of music reinforcement dictate all future trends, we acknowledge that the areas of motion picture sound and speech reinforcement are holding their own. New HF horn designs with skewed patterns provide smooth coverage over large areas and thus simplify array design. Moving in the opposite direction, tapered line arrays of small direct radiator loudspeakers are offering some remarkable solutions to complex coverage problems.

Finally, we are pleased to note the impact of the microcomputer on our industry. The base of measurement has never been broader than it presently is, and current methods of modelling arrays and coverage are just getting underway. ■



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Ask your Ampex distributor, or send the coupon, for an AG-500 demonstration—measured to your needs.

New Products and Services

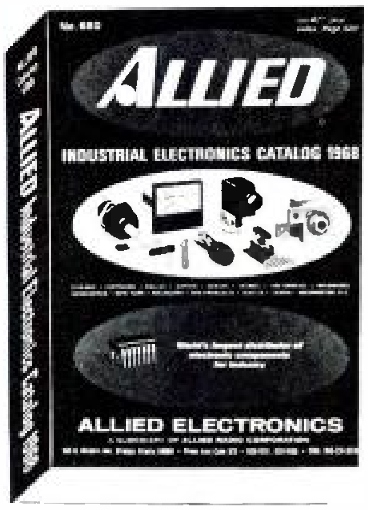
Sixteen Years Ago

Audio Control Center

The model AC-155 offers a combination of compactness, mobility, appearance, and strength, plus technical features of professional quality. The four-foot-wide unit has 14 inputs and accommodates six low-level and eight high-level audio sources. An all-channel cue system, muting, monitor amplifier and speaker are included. Also part of the console are two turntable/arm/cartridge systems. Accessories include a matching bench/lid which gives protection to the console in transport and becomes a seat for the operator on location. Tape cartridge systems and a cardioid microphone are also available.
Mfg: Sparta Electronic Corp. Price: On request

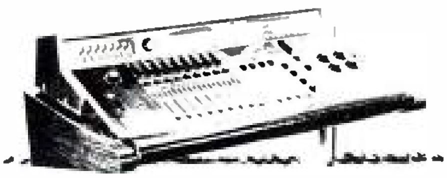
Industrial Catalog

This wopper of a catalog has over 600 pages with over 50,000 stock items listed. More than 500 manufacturers are listed. The listings show prices for purchase in various quantities of every type component, including IC's, semiconductors, vacuum tubes, relays, times, transformers, resistors, capacitors, connectors, soils, chokes, sockets, plugs, jacks, switches, fuses, batteries, clips, lamps, wire, and cable. Other sections show test instruments, two-way radios, recording equipment, sound equipment, power supplies, hardware, etc. There is a full index.
Mfg: Allied Radio Corp. Price: No charge



Modular Console, Model 9200

This is a basic, unitized enclosure for modular assembly. A full line of components may be selected to meet the specific requirements of a particular installation. As many as 27 strip modules may be installed, along with up to 4 VU meters. Each module is hinged forward for easy access. Each input panel may contain a rotary mixer, straight-line mixer, equalizer, echo send attenuator, and a multi-switch assembly. Pan pot panels, talk-back panels, playback and monitor panels, jack strips, graphic equalizers, and the meter assembly are all variable to individual requirements.
Mfg: Altec, Div of Ling Altec, Inc. Price: List available on request



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New Products

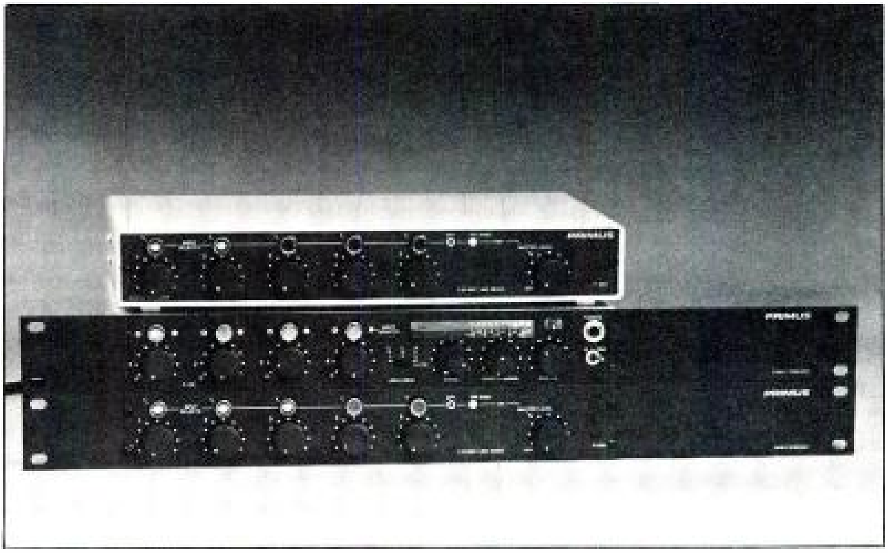
DUAL CHANNEL AUDIO ANALYZER

• RE Instruments' model RE201 Dual Channel Audio Analyzer utilizes an FFT analysis process and a CRT display to provide a completely digital instrument designed for rapid, repeatable, and accurate testing of radios, amplifiers, and most audio equipment. The measurement capabilities include Total Harmonic Distortion (to -90 dB), intermodulation, transient intermodulation distortion, difference frequency distortion (to -70 dB), AC levels (up to 75 kHz), DC levels, phase, frequency, separation (to -80 dB), and wow-and-flutter. Each of these 10 basic measurements can be defined to be performed in 10 different ways. The RE201 gives the user non-volatile storage capability for up to 90 such measurement definitions. It also allows the user to define and store multi-measurement sequences. These may be recalled with a single keystroke to simultaneously display up to nine two-channel measurement results on the CRT. The RE201 is IEEE interfaceable; it also has an RS232 port for mass storage or hard-copy documentation. The unit has a modular design utilizing two individual 16-bit microprocessors. Extensive EMI shielding includes a separately shielded IEEE interface port, digital section, and CRT section. A full range of plug-in options for increased capability will soon be available.

Mfr: RE Instruments Corporation
Price: $14,465.00

Circle 53 on Reader Service Card

AUDIO MIXERS



• Ramko's P-4M and P-5MX mic/line mixers are the latest addition to their Primus audio group. Both are offererd in mono and stereo versions and are available in 1¾-inch table-top or rack mount configurations. The P-4M provides four mixing channels and six balanced inputs with selectable high/low shelving equalizers for channels 1, 2, or all. Other features are selectable Peak VU solid-state meter ballistics, phone driver, phones, master and monitor controls, and cue on all inputs. The P-5MX is designed to function as both an expander for the P-4M (which combined will provide 11 inputs and 9 channels), as well as a stand-alone five-channel mixer with send/receive on each channel. Both units feature XLR-type connectors, balanced inputs and outputs, and gain select on all inputs (mic thru +26 dBm, S/N of -83 dB). Distortion is .008 percent and frequency response is 10 Hz to 20 kHz, +0, -1.5 dB. All units utilize conductive plastic controls, long-life switches, and are covered by a three year warranty.

Mfr: Ramko Research

Circle 55 on Reader Service Card

HIGH PERFORMANCE MULTIMETERS

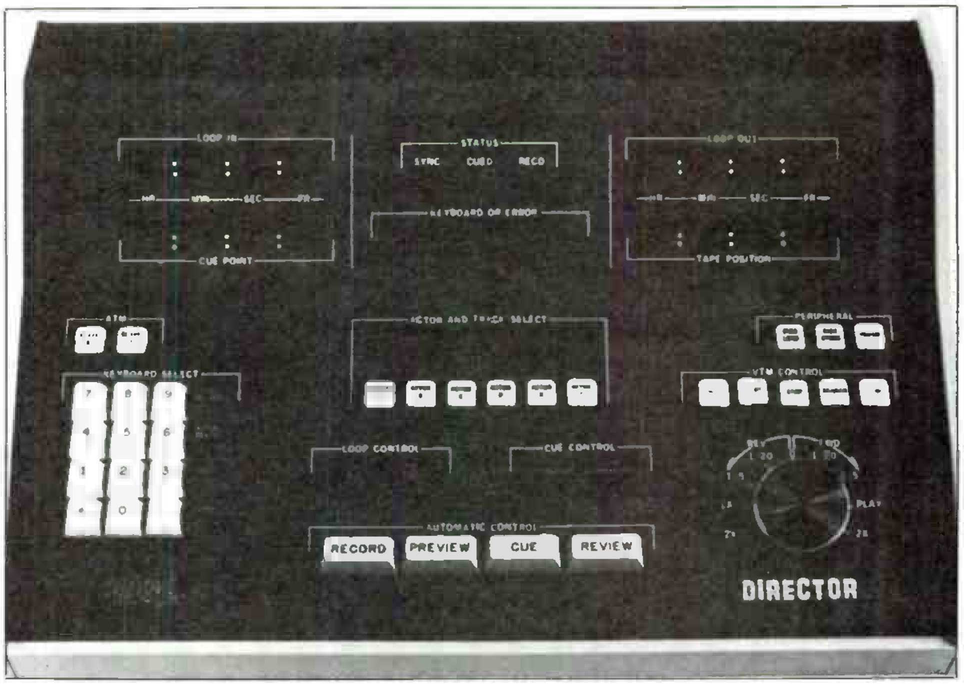
• The Fluke 70 Series of hand-held multimeters combines features such as a single eight-position switch for function selection, and both digital and analog displays. A simple analog bar graph moves up and down a 32-segment scale, updating 25 times per second—10 times faster than the digital display. At a glance, the user can note the trend indications for peaking and nulling measurements or continuity checks. The 3200-count digital display gives the Fluke 70 Series the same resolution as a typical 4½-digit multimeter for displays above 2000 counts. When measuring a 220-volt line, a 24-volt power supply, or a 20-milliamp current loop, the increased count provides an extra digit of resolution over traditional 2000-count meters, which must change ranges for these measurements. A unique auto-ranging function instantly selects the correct measurement range. All functions are color-coded and clearly designated on the liquid crystal display. All three Fluke 70 Series models measure DC voltage to 750 V, AC voltage to 1000 V, current to 10 amps, and resistance to 32 megohms.

Mfr: John Fluke Mfg. Co., Inc.

Price: Fluke 73: $ 85.00
Fluke 75: $ 99.00
Fluke 77: $129.00

Circle 56 on Reader Service Card

AUDIO-FOR-VIDEO LOOPING CONTROLLER



• United Media's DIRECTOR and TRANSLATOR are microprocessor-based looping controllers employed in the process of Automatic Dialogue Replacement (ADR). Unlike the conventional synchronizer, these products utilize the same concepts of tying audio to video with full on-line editing capability as their counterparts in film production. The DIRECTOR handles up to six voices separately, routing each performance to its own track. Functions such as high-speed "rock and roll" of tape and picture, storage of loop input and output points, and recording of each cue point as an automatic re-assembly point for editing are standard. The entire ADR can be pre-programmed using frame by frame accurate SMPTE time code cueing with single keystroke control. All data is stored on floppy disk for backup. The operator can, through the captive memory, scroll back through for cue review, or play through the tape. Both audible beeps and visual wipes are produced. The DIRECTOR can also be equipped with CRT video display and/or high-speed printer for expanded review capability. Talent may perform work at separate times being individually cued for start points and pacing. Video and audio cues generated by the DIRECTOR can be varied in terms of pre-promt timing depending on individual talent reaction time. Standard interfaces accommodate one VTR and one ATR as the minimum configuration and expand to handle two or more VTRs, ATRs, or a combination of the two, depending on the nature of the work to be performed.

The TRANSLATOR incorporates only the DIRECTOR's functions required for recording cues. The TRANSLATOR allows the operator to pre-program the production via off-line cue recording. Presence at the final session or sessions becomes optional. Interfaces for floppy disk, paper-tape punch and reader, and either high speed printer or TTY are standard for both systems.

Mfr: United Media

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

People, Places

• **Almon Clegg** has been appointed to the position of general manager of the Audio Division and Communications Division at the **Matsushita Technology Center** (M-TEC). The promotion, announced by **Adam Yokoi**, vice president of M-TEC, gives Mr. Clegg expanded responsibilities in the areas of product research and business development in high technology fields. Mr. Clegg joined Panasonic in 1974 as manager of the Audio Engineering Department. He was promoted to the position of assistant general manager of the Product Engineering Division in 1978 and assumed a similar post when M-TEC was formed in 1982. Prior to coming to Panasonic, Mr. Clegg worked with **General Electric** and was an Associate Professor of Electronic Engineering at **Illinois State University**. Mr. Clegg is a Fellow of the **Audio Engineering Society** and a Senior Member of the **IEEE**.

• Following the success of the US roadshow in September of this year, **Solid State Logic** is taking to the road again. Starting on November 12th the roadshow will be visiting major cities in Germany, Portugal, Spain, Belgium and Holland, returning to the UK just before Christmas. SSL's chief marketing executive for Europe, **Jay Denson**, will be leading a team of engineers to visit broadcasters and recording studios in these countries. On show will be a 32 channel SL6000 E Series Stereo Video System integrating a synchronized three machine (Video, Studer A800 24 track and A810) set-up with Cart machines controlled via SSL's Events Controller. The emphasis is on providing hands-on working demonstrations to enable would-be purchasers to get a thorough close-up look at one of the world's most advanced and versatile video-post production broadcast console. A specially prepared Audio-Visual presentation will be used to explain the philosophy and applied techniques of the system.

• Saturday evening, October 8th, **Peter Eros** conducted the **Peabody Symphony Orchestra** in the first concert in the newly renovated **Miriam A. Friedberg Concert Hall** in the historic **Peabody Conservatory**, with works by Hugo Weisgall, Verdi, Weber, Bizet, and Beethoven. Soloists included **Myra Merritt**, Soprano; **Richard Cassilly**, tenor; **Ellen Mack**, piano; **Berl Senofsky**, violin; and **Stephen Kates**, cello. Reviewers and faculty commented on the improved "sonic warmth" given by the increased reverberation time, use of all hard materials, and movable stage-end panels that were incorporated in the renovation process. The remodeled hall can accommodate opera, with the movable panels rotated to their "teaser/tormentor" positions, and with the adjustable pit uncovered. The hall also features complete recording facilities. **Klepper Marshall King** were acoustical consultants for the renovation, with **Jewell Downing Associates** of Baltimore as architects. The new sound reinforcement system for the Hall, used primarily for speech and motion pictures, features a horizontal line-source loudspeaker system, using thirty-two **JBL** eight-inch loudspeakers. Because of the rectangular shape of the Hall, the loudspeaker system operates as an infinite line with good uniformity of coverage and a very inconspicuous appearance. Supplementary delayed loudspeakers are used for listeners in the balcony and underbalcony areas that lack line-of-sight to the main proscenium line-source system. The sound reinforcement system and the recording system were joint designs of **Alan Kefauver**, Recording Studio Director, and **Dave Klepper** of KMK. The systems were installed by **Recording Consultants, Incorporated**, of Silver Springs, Maryland, with **Dave Paseur** and **Ed Johnson** responsible for most of the installation. All loudspeakers throughout the hall, studio, and control room are JBL (including bi-radial control room monitors); all amplifiers and most equalizers are **UREI**; delay and reverberation accessories are **Lexicon**. The recording console is a **Sound Workshop** Series 1600; the reinforcement console is a **Yamaha** 916.

• Three **3M** 32-track digital mastering systems have been sold to major recording studios in Nashville, announced **Tom Kenny** of 3M's Magnetic Audio/Video Products Division. Two units were purchased by **Norbert Putnam** of the **Bennett House**; the third by **Bill Roach's Castle Recording Studios**. According to Kenny, 3M is definitely in the digital recorder business, rumors to the contrary. The rumors began, he said, when there was a drop in the digital system price coming around the same time 3M divested itself of the analog recorder business. The company's digital recorder system was introduced almost six years ago.

• **Litton Industries and Quad Eight Electronics** have announced that Quad Eight has purchased from Litton its **Westrex Sound Recording** operation in the United States and all Westrex operations in the United Kingdom. The merged businesses will operate as **Quad Eight/Westrex**. Westrex, originally the motion picture sound equipment division of **Western Electric**, was instrumental in the development of sound for motion pictures over 55 years ago, and has continued to supply the highest level of equipment to studios all over the world. The Westrex operations were acquired by Litton in 1958. Although Quad Eight has always sold at least 50 percent of its products internationally, as Quad Eight/Westrex it will now have a multi-national manufacturing base with the acquisition of Westrex's manufacturing facility in the U.K. The company will manufacture both recording equipment and consoles in the U.S.A. and the U.K., allowing a greater worldwide distribution system for all of its products.

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How to Build a Better Compact Professional Recorder

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