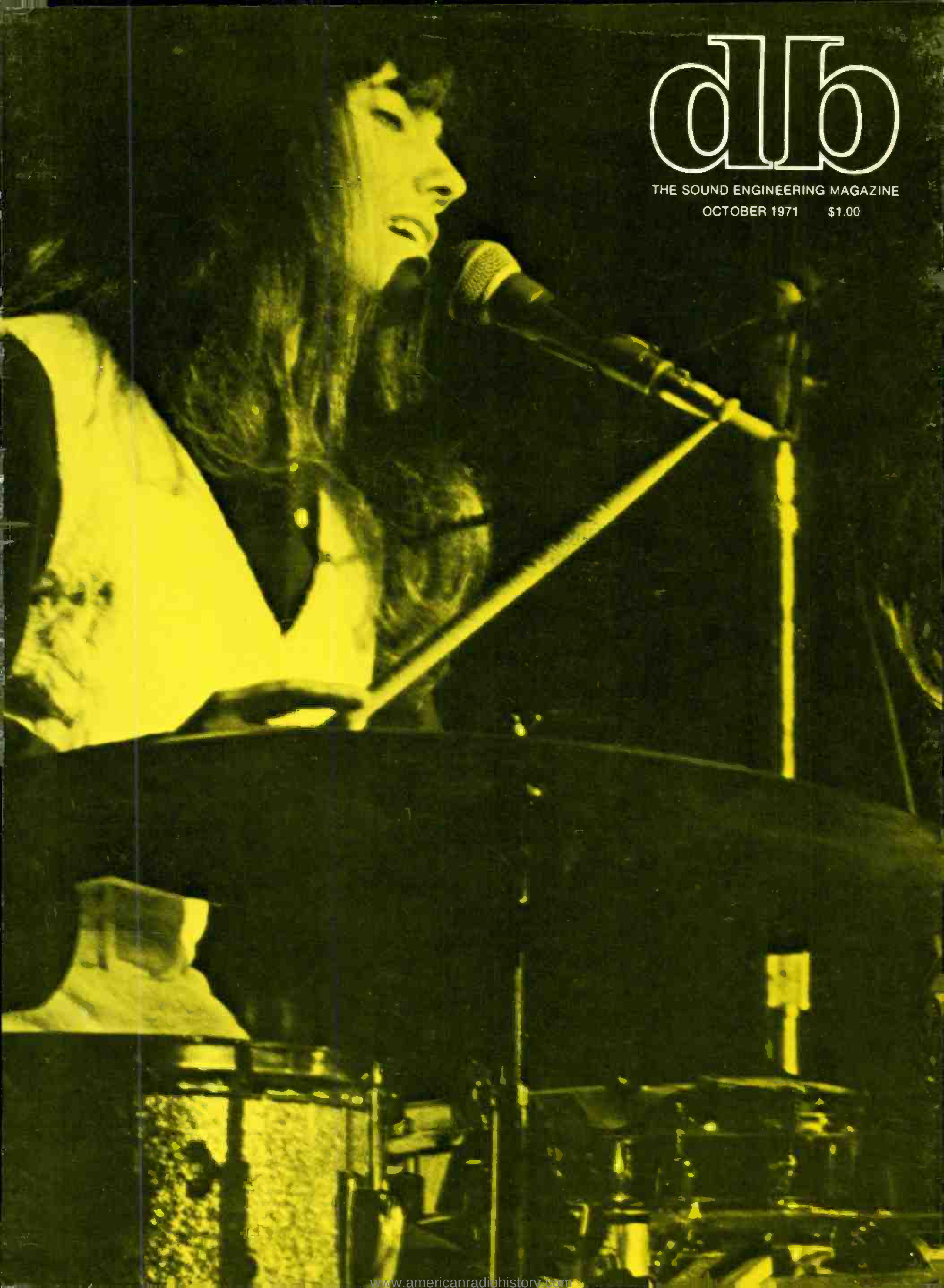
# db

THE SOUND ENGINEERING MAGAZINE

OCTOBER 1971 $1.00



# COMING NEXT MONTH

• Our tape recorder was present when we sat down a group of nine leading independent studio owners and engineers. Their frank opinions on what it takes to make a profitable operation, as well as many facets of studio operation, make valuable reading you will not want to miss.

Cameras got poked into a variety of recording studios and pictures resulted that will show you how some winning studios got that way. You may just see some ways in which you can make your operation better. One studio is on Long Island, New York; one is in Southern California; and one is in Canada.

And there will be our regular columnists: George Alexandrovich, Norman H. Crowhurst, Martin Dickstein, Arnold Schwartz, and John Woram. Coming in **db**, The Sound Engineering Magazine.

# ABOUT THE COVER

• The world of rock sound has its own particular public address problems. They are totally different from those of any other sound-reinforcement needs. See R. H. Coddington's article on page 30. Photo is courtesy of Shure Bros., Inc., whose microphone is being used.


# db

THE SOUND ENGINEERING MAGAZINE

OCTOBER 1971 VOLUME 5, NUMBER 10





db is listed in **Current Contents: Engineering and Technology,**





db, the Sound Engineering Magazine is published monthly by Sagamore Publishing Company, Inc. Entire contents copyright © 1971 by Sagamore Publishing Co., Inc., 980 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 $6.00 per year ($7.00 per year outside U. S. Possessions, Canada, and Mexico) in U. S. funds. Single copies are $1.00 each. Controlled Circulation postage paid at Harrisburg, Pa. 17105. Editorial, Publishing, and Sales Offices: 980 Old Country Road, Plainview, New York 11803. Postmaster: Form 3579 should be sent to above address.

# letters

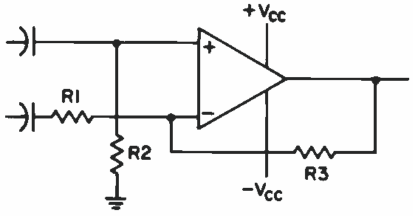



FIGURE 1.

The Editor:
There is a small discrepancy in Walter Jung's article *A Differential Bridging Amplifier* (July '71 **db**). The output into 600 ohms (24-volt supply is not +22 dBm but slightly lower. Its correct value may be computed as follows.

20 volts peak-to-peak is 10 volts peak, or $10/(2^{1/2}) = 7.07$ volts rms. Since *Power* = (*volts*rms)$^2$/*resistance*, we calculate the power as $7.07 * 7.07/600 = 50/600 = 0.083$, or 83 milliwatts. Since 0 dBm is 1 milliwatt, our power in dBm is $10 * \log(83/1) = 10 * \log(83) = 19.2$ or +19.2 dBm. This is half the power quoted by Mr. Jung (a loss of 3 dB is a 50 per cent power loss).

Granted, it's a small point. But if you start putting pads on the output to decrease the power (and noise) for driving another stage without overloading, you only have half your power. A 100-watt auditorium amplifier will only be delivering 50 watts, and a 10-watt cutter-head amp will only deliver 5. Use of the amp (and I must admit I never could have designed it) should be calibrated accordingly.

The four diagrams will interest all readers of Mr. Jung's article. The first FIGURE 1 shows his circuit idea, but using an integrated circuit operational amplifier (perhaps a Motorola MC-1709). R2 should equal R3 for a zero offset opamp. Gain equals R3/R1, and input impedance equals R1 plus R2.

FIGURE 2 is for increased output power. The class A output transistor is located within the feedback loop to cut distortion, although this does create a −0.6 volt output offset. Due to negative feedback, this reflects itself as a +0.6 volt offset at the i.c. output and thereby limits the positive peak to 0.6 volts less than it was designed for.

FIGURE 3 shows a method of eliminating this offset, although I haven't personally verified that it works. The diode's voltage drop offsets the emitter

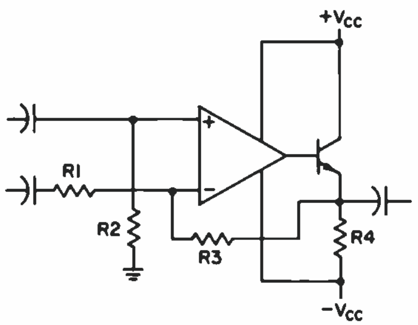
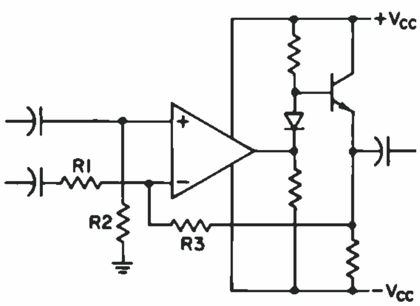



FIGURE 2.

FIGURE 3.




## advertisers index




db

THE SOUND ENGINEERING MAGAZINE

## SALES OFFICES

**New York**
980 Old Country Road
Plainview, N.Y. 11803
516-433-6530

**Dallas**
Roy McDonald Associates, Inc.
Semmons Tower West
Suite 714
Dallas, Texas 75207
214-637-2444

**Denver**
Roy McDonald Associates, Inc.
846 Lincoln Street
Denver, Colorado 80203
303-825-3325

**Houston**
Roy McDonald Associates, Inc.
3130 Southwest Freeway
Houston, Texas 77006
713-529-6711

**Los Angeles**
Roy McDonald Associates, Inc.
1313 West 8th Street
Los Angeles, California 90018
213-483-1304

**Portland**
Roy McDonald Associates, Inc.
2305 S. W. 58th Avenue
Portland, Oregon 97221
503-292-8521

**San Francisco**
Roy McDonald Associates, Inc.
625 Market Street
San Francisco, California 94105
415-397-5377

# ELECTRIC LADY and MEDIASOUND can now record silence as well as sound. (Both have the Dolby System on 16-tracks)

## $30,000 worth of session is worth keeping quiet.

It's easy to record sound; any studio can do it. It's just as easy to add noise to the music you're recording; whether you like it or not, the tape itself does that. To make matters worse, tape noise increases by about 10 dB when sixteen tracks are mixed down to two.

At Mediasound or Electric Lady the Dolby System puts you 10 dB ahead of the quietest tape money can buy. Print-through, crosstalk, modulation noise and even some distortion components are all reduced, quite apart from the dramatic reduction of tape hiss.

You'll get better technical quality in your next New York recording by calling:

| | |
|---|---|
| **ELECTRIC LADY** | **(212) 777-0150** |
| **MEDIASOUND** | **(212) 765-4700** |
| **DOLBY LABORATORIES INC.** | **(212) 243-2525** |

AND "DOLBY" ARE TRADE MARKS OF DOLBY LABORATORIES

*Circle 13 on Reader Service Card*

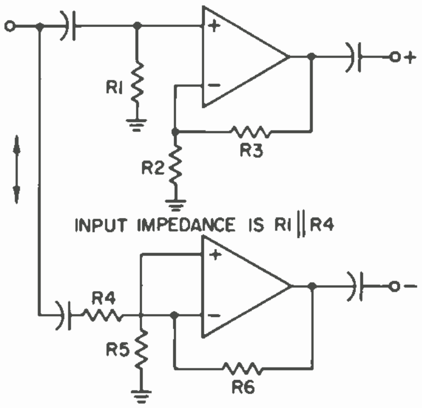



FIGURE 4.

in the opposite direction; thus the voltage at the emitter is the same as at the i.c.'s output. Of course, if the i.c. delivers enough power to do the job, the class A stage is unnecessary.

FIGURE 4 shows a transformer-less output stage, using two operational amplifiers. Since the outputs are out of phase, the circuit can drive a balanced line. Phase reversal is simple; a reversing *dpdt* switch could be inserted, just as with a transformer.

Class A output stages (discussed above) should be added if the i.c.'s don't have enough power to drive the line. Stage gain can be set quite high; by setting R1=R3 and R5=R6 (for proper bias), eliminating R2 and setting R4=R6 yields unity voltage gain (providing R1 is large enough for your i.c.).

Care should be taken to have the stage gains equal. For experimenting, I used a Motorola MC1303 (dual op amp in a single package), but I could only get the noise figure with this unit down to about −65 dBm with a +8 dBm output (a class A output driver was used).

I must admit that, unlike author Jung, I'm partial to split supplies. The studio I'm attempting to build from scratch in my basement is supposed to operate off one dual supply of plus and minus 28 volts, with each unit (19" rack mount) containing further reduction when required. If ground loops do crop up, the transformerless balanced line will probably be the cheapest corrective action (providing the extra noise from the active components doesn't create a problem).

*Peter Dworsky*
*Scarsdale, N. Y.*

*Mr. Jung responds*:
To answer Mr. Dworsky's comments on the power output of the DBA circuit of FIGURE 2 (page 23 of July issue), I must offer an apology for lack of clarity on this point. The text refers to a 20 volt peak-to-peak level which Mr. Dworsky has interpreted as the maximum output capability.

Were it so, the circuit would be limited to the +19.2 dBm mentioned. Actually the circuit will deliver more than 22 volts p-p with a 24 volt supply, and this is a little over +20 dBm. The specification should read 22 *volts* or +20 dBm and not the +22 dBm mentioned. I regret any confusion this may have caused.

To shed a little more positive light on the situation, let it clearly be stated that there is no magic limit to the +24 volt supply potential. Power output of this output stage is limited by 2 factors, standing current from Q11 (set by R19), and supply voltage. The circuit will actually deliver the full supply voltage minus a volt or less saturation drop for Q10-Q11 and the same for Q9-Q12. In practice this means the supply minus 2 volts or less, which is downright efficient. For larger peak levels into 600 ohms, decrease R19 correspondingly, and raise supply voltage. For instance with R19 at 25 ohms and a supply of 30 volts, +22.3 dBm was measured out of this amplifier. Readers who may want to use the circuit might note that these different supply voltages may be used with only readjustment of R14 for symmetrical clipping.

As far as the operational amplifier configurations submitted by Mr. Dworsky are concerned, it is certainly true that the DBA concept can be implemented using 709's, 741's or other standard chips. But the main intent of the article was to: 1) present the concept in a circuit realization which simultaneously optimized as many significant factors related to practical use as possible. These are; high-in, large common-mode rejection, high input capability, single resistor gain control without sacrificing other performance specs, efficient biasing through current source usage, very high linearity, symmetrical clipping with fast overload recovery, direct coupling for flat response and low phase shift, and large linear output swing for line driving capability. All of the above are features of this circuit. No currently available i.c. can provide these criteria simultaneously; thus the discussion on a hypothetical chip which would. This is 2), the second major point aimed at by the article-how much simpler life could be if audio-oriented, audio performance-optimized i.c.'s were available to fulfill the needs of professional audio engineers. A good example of this would be Mr. Dworsky's FIGURE 4, which while possible with 2 op amps, is best done within a single circuit. Which could just well turn into another **db** article. . . .

# See if you can walk away unaffected.

If you ever have an opportunity, sit down at the controls of an Olive console and play a tape through it. The flexibility and experimental freedom may be overwhelming when you realize all of the features you have always wanted in a console are at your fingertips.

Surrounded by human-engineered, technically sophisticated equipment that is this easy to operate, you enter a new dimension of confidence. Setup is fast and easy with immediate response to your touch plus visual displays that provide instant feedback on complete console status.

Try an experiment: pick out an instrument on the tape and enhance it with our unique four-section equalizer. Now you can begin to understand why everyone's talking about it. If you like this kind of excitement then play the master tape back and watch the Remix Programmer duplicate your manual mix . . . automatically.

Further modification is very simple and the last recorded "memory" is permanently stored right on the original master. Amazing? This console is designed for the computer age.

Stop by for the "Olive experience" at Booths 70, 71 and 72 during the AES Convention in October. It may have an effect on you. If you would like further information on the Olive Series 2000 Automated Console or the smaller Series 2500, contact our US distributor or write direct. Either way your request will receive personal attention.

olive
WHENEVER THERE IS A NEED

OLIVE ELECTRO DYNAMICS INC.
2570 Paulus
Montréal 386, Québec
Canada
(514) 332-0331
Cable Olivel, Montréal

US Distributor
WESTLAKE AUDIO INC.
6311 Wilshire Boulevard
Los Angeles, California
90048
(213) 655-0303

*Circle 19 on Reader Service Card*

George Alexandrovich

# THE AUDIO ENGINEER'S HANDBOOK

## Gain Shifting in Two-Way Communications

• Have you ever wondered why it is hard to interrupt a person on the phone you talk to who is across the country? Have you noticed that if you talk continuously you don't hear him say a word? Electronically-minded people are aware that some sort of squelching is going on. It is used to reduce or eliminate disturbing echo effects which existed in the old telephone lines. This echo was produced by leakage of the signal being received into the outgoing mic circuit because of imperfect balance of the hybrid or repeater coil, or through the acoustical coupling between the receiver and mic. Delay of the sound produced by the long telephone lines allowed the talking person to be hearing himself a fraction of a second later—which is very confusing to him. The type of squelch presently used in long distance telephone lines, all sorts of limiting, compression, suppression, or expansion are nothing else but a gain change automatically produced in order to eliminate or minimize adverse effects or conditions existing in the communication loop.

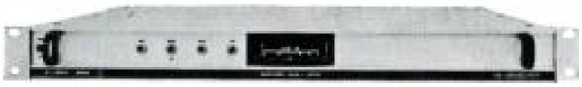
These types of gain controls, our subject for this column, are all automatic and are soft switches—in contrast to circuits used for data transmission over the limited capability transmission lines which are hard switches with fast rise and fall switching characteristics. Soft switches are designed to accomplish the transition from one level to another inconspicuously fast and smooth.

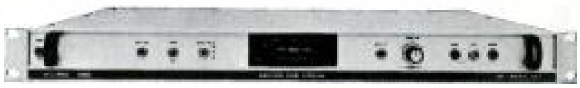
We are all familiar with conventional methods of changing gain in limiters and compressors to prevent overloads and decrease the dynamic range. In these circuits gain is being reduced as the level rises. Opposing the compressor and limiter are expandor circuits which *increase* the gain of the circuit as the level increases. *Both* types of circuits are the heart of gain-shifting communication system we will discuss.

Consider that you have to set up two-way communication between two locations, using microphones and speakers with necessary amplifiers and hardware. There will be two nets—one carrying signal from location A to location B and the other from B to A. A mic at location A picks up the signal and feeds it into the amplifiers for reproduction at location B. A mic at location B, being in the same general area, inevitably "hears" this reproduced signal and feeds it back to location A. If gain in both paths is high enough, the same signal will circulate in the system being amplified until it will bloom into one continuous squeal called acoustical feedback. If a person in room A wishes to communicate with the person in the room B when the gain of the system is at a safe level (below the feedback level) reproduce levels at both locations will be too low for comfortable listening. Every increase of either circuit will result in oscillations. From this it follows that we have limitations on gain in both directions, and we shall refer to this as *critical loop gain.* Maximum loop gain depends greatly on the mic-speaker coupling as a function

# shhhh!

DON'T MAKE A SOUND UNTIL YOU HAVE AUDIMAX AND VOLUMAX.





This is the team that quietly goes to work to produce a perfect sounding program. Audimax eliminates distortions like thumping, audio holes and the "swish-up" of background noises. Volumax prevents overmodulation and limits program peaks, permitting broadcasters to achieve maximum power from each watt of carrier power. Together, Audimax and Volumax produce a new excellence in sound control, increase audience coverage and amplify your station's profits.

**CBS LABORATORIES**
A Division of Columbia Broadcasting System, Inc.
227 High Ridge Road, Stamford, Connecticut 06905

*Circle 36 on Reader Service Card*

A & B Duplicators Ltd.
A & R Recording
Accurate Sound Corp.
Aengus Enterprises
Alembic Labs
Allegro Recording
American Amplifier & Television
American Broadcasting Co.
American Foundation for the Blind
American Gramophone
American Lutheran Church
American Telephone & Telegraph
Ampro Corp.
Aquarius Sound
Atlantic Records
Audio Acoustics Equip. Co., Inc.
Audio Graphic Labs
Audio & Electronic Consulting
Audio Finishers, Inc.
Audiotronics Inc.
Apostolic Studios
Artcraft
Richard Becker Enterprises
Bell Telephone Labs
Belt Sound
J. T. Bernard Studio Inc.
Bizarre Inc.
Bose Corp.
Bushnell Electronics
Caedman Records
Capitol Records
CBS Electronic Video Recording
CBS Laboratories
CBS Ltd., London
CM Labs
Columbia Records
Command Studios Ltd., London
Compagnie Phono. Francaise, Paris
Criteria Recording Inc.
Continental Recording, Inc.
Creative Visual Sonic Eng.
Custom Fidelity Co.
Dage Corp.
Danmarks Radio, Soborg, Denmark
Data Packaging Corp.
Dax Company
D B Audio Corp.
Delane Lea Music Ltd., London
Delaney & Bonnie
de Medio Engineering
Digiac Corp.
Drum Hill Communications Group
Duquesne University
EAB Studios
Eastern Sound Services, Toronto
Eastman School of Music
Echo Sound Studios Inc.
8 Track Recording Co.
Electra Sound
Electric Lady
Emperical Sound
EUE/Screen Gems
Fedco Audio
L. P. Graner, Inc. Engineers
GRT East

# is your name on this list?*

Grief Studios
Hagens Recording Studio Inc.
Hanley Sound
Wally Heider Recording
The Hit Factory
Holland Dozier & Holland
Illinois Bell Telephone Co.
Impact Sound Recording Studios
Intermedia Systems Corp.
International Electro Acoustics
International Telecommunications
J T Sound
Jackson Sound
Jefferson Airplane
Jewel Recording
K P P C
K V M L
Kelmar Systems
Keysor Century Corp.
Kingsley Sound
Knight Recording
Larabee Sound
Lewellyn & Martin, Inc.
Location Recorders
London Records
M C I
M I T
Magnetic Recording Systems Inc.
Magno Sound
Dennis Maitland, Inc.
Manhattan Audio Co., Inc.
Manta Sound

Marlboro Engineering Co.
Media Sound
Mercury Sound Studios
Midwestern Sound
Mitre Productions, Ltd.
Monumental Films
Music Factory, Inc.
National Public Radio
National Recording Studios
Natural Sound
Norsk Rikskringkasting, Norway
Northern Motion Picture, Toronto
Northwest Audio Development
P B S Studios, Ltd.
Photo Magnetic
Post Horn Recording
Quadraphonic Sound
RCA Records
RCA Records, Engineering Developm't
RIC Studios
Record Plant
Recording Studios, Inc.
Reeves Sound
Regent Sound
Riker Information Systems, Inc.
Rodgers Organ Co.
Rosner Custom Sound Service
Ryder Sound Services Inc.
Rossi Enterprises
Seco Labs Inc.
Robert Slye Electronics, Inc.
J. H. Sparks, Inc.
Sound Arts Co.
Sound 80 Inc.
Sound Ideas Studio
Sound Specialties
Streeterville Studios
Studio Center
Studio 236
Studio 70
Studio Techniques, Paris
Suburban Sound
Syncron Sound Studios, Inc.
Theatre Editing Studios
Theatre Sound Inc.
True Sound & Music
United Recording of Nevada
United Recording Studios
University of Illinois
University of Missouri
University of North Dakota
U. S. Dept. of Interior
U. S. Marine Corps
United Research Labs
Vanderbilt Museum Commission
Vantone Recording
Vicount Records
Video Film
Visual Data
W B A I
W C C M
W G B H
W H Y L
W J I B
W K C R

*Continued* ▶

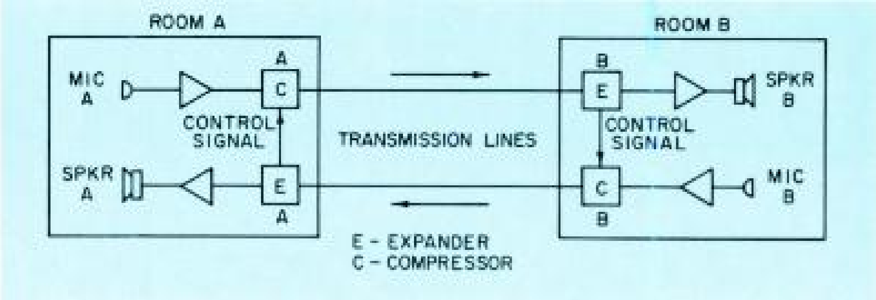



*Figure 1. Signal travelling from mic A triggers expander B. This, in turn, triggers compressor B. Signal is then reproduced through speaker B. If there is another signal present (which is triggering expander A and thereby compressor A) then level in the mic A line is suppressed sufficiently, preventing triggering of expander B. However, mic A signal will still be heard at the quiescent level through speaker B.*

of the directional property of the transducers (speakers and mics) and of the acoustics of the room.

From the above we can deduce that the only remedy to the situation is to raise the gain in the channel which carries the original signal, while lowering to in the line in the opposite direction. Going to extreme, we can momentarily turn off the return channel while raising the outgoing line gain as much as we desire, without the ill effects of feedback. Theoretically this is quite acceptable, however, practice tells us differently—and here is why. Referring back to our first statement, about interruption—it may be just as disturbing to not be able to interrupt or interject (in the course of normal conversation) as hearing your own echo. Interjection capability is most desirable, even if one has to raise his voice slightly in order to be heard. This gives more vivid coloration to the communications, with less strain on the participants who often forget that they are in different rooms, sometimes in different cities or even countries.

As a result of many years of practical evaluation of the principles and systems, as well as different acoustical environments, the following rules can be set up.

A successful two-way open-mic system will function well if the following conditions exist:

1. Gain of the entire system is set so that no acoustical feedback is present and varied in both circuits in such a way that the sum of both gains in the two loops always remains constant. This is accomplished by using compression of the unused channel while expanding the gain in the working channel. For best results and maximum levels the relative amounts of gainshifting should be closely matched.

2. Gain of either channel should be permitted to increase only to the point that a person standing three feet in front of the mic will be heard at the other location at the level resembling his own voice as if he was standing in place of speakers and talking at normal level. This can be easily verified using a sound-level meter.

3. The more directional are the mics and speakers, and the less acoustical coupling exists between the two at each location, the more quiescent gain the system can have before gain shifting is introduced. Then, in order to achieve acceptable listening levels, less gain shifting is required and the system sounds more natural—with more interjection capability since less expansion and compression is required. This will explain why telephone lines have such a high degree of gainshifting and why it is hard to make communications sound pleasant.

The importance of having good acoustical isolation between the mic and speaker can be demonstrated if the system is assembled in an anechoic chamber (where there are no reflections from the walls and gain can be raised to the desired level without the fear of feedback). This also underlines the importance of the acoustical treatment of the room, since it reduces reflections and suppresses room resonances. This takes us to the next requirement.

4. If there are resonant peaks in the acoustics of the room then reducing them through tailoring the response of the amplifying equipment (environmental equalization) would increase the possible system gain before feedback. Again this would reduce the needed amounts of gain shifting.

5. Speed with which the gainshifting has to occur should be in the vicinity of a couple of milliseconds. *Instantaneous* response is *not* desirable since triggering of the gainshifting circuits would happen on every fast transient click or tap. Too slow a triggering will result in the loss of

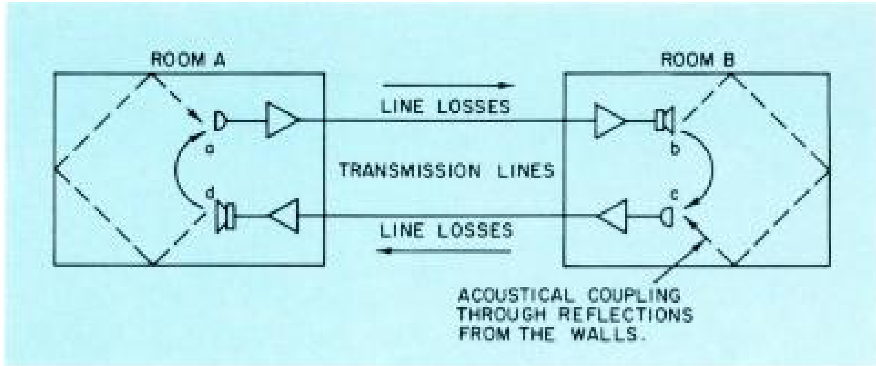



*Figure 2. The loop gain of a transmission system. ab + cd = loop gain; da and bc are acoustical coupling which complete the loop.*

some of the parts of the words. Speed of the expander and the compressor action should be identical, since momentary increase or decrease of the total system gain will result in ringing if the sum of the gains gets higher. Softer reproduction will result if the sum of the gains gets lower. This requirement sometimes is overlooked, since in testing the system gain, readings are taken with steady tones—yet dynamic balance of the gains is one of the most important requirements.

6. The less gain shifting is used, the more natural the system sounds. In the average conference room, with a moderate amount of sound treatment of the room and using directional transducers, the amount of gainshifting should be around 15 dB. This change of gain will achieve adequate levels so that a person can be talking in a normal tone of voice anywhere in the room and be heard at the other end as if he was there himself.

7. Threshold of the gain shifting action should be adjustable and be set to trigger a few dB above the ambient noise of the room. Also, precaution should be taken so that the triggering doesn't occur from the sounds reproduced by the speaker reproducing sound arriving from the other location. For this purpose, a special lockout circuit is built into the electronics so that (although interjection capability exists) the gain shifting trigger circuit is disabled in the presence of the incoming signal. Therefore, the microphone, when it "hears" the speaker-reproduced signal, doesn't actuate the suppressor circuit for the incoming line. This circuit is basically a compressor working in the triggering circuit and actuated by an incoming signal.

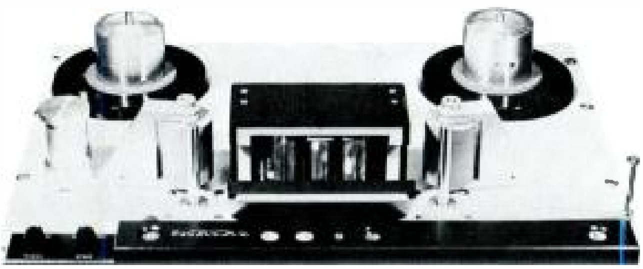
What has been described here is a system consisting of two stations and two sets of electronics. Much more complex systems have been built and are now in operation using this principle of gain shifting—connecting several cities across the country. At each location, there are various-size conference rooms, each connecting into one common conference loop working through conventional dial telephone lines. At each location, many people can participate in the conference and be heard at all locations or cities.

Training simulators for the moon missions at NASA in Houston have been using the same communication equipment, as well as many other organizations needing the ability for groups of people to communicate without using any push-to-talk switches found in the conventional intercoms.

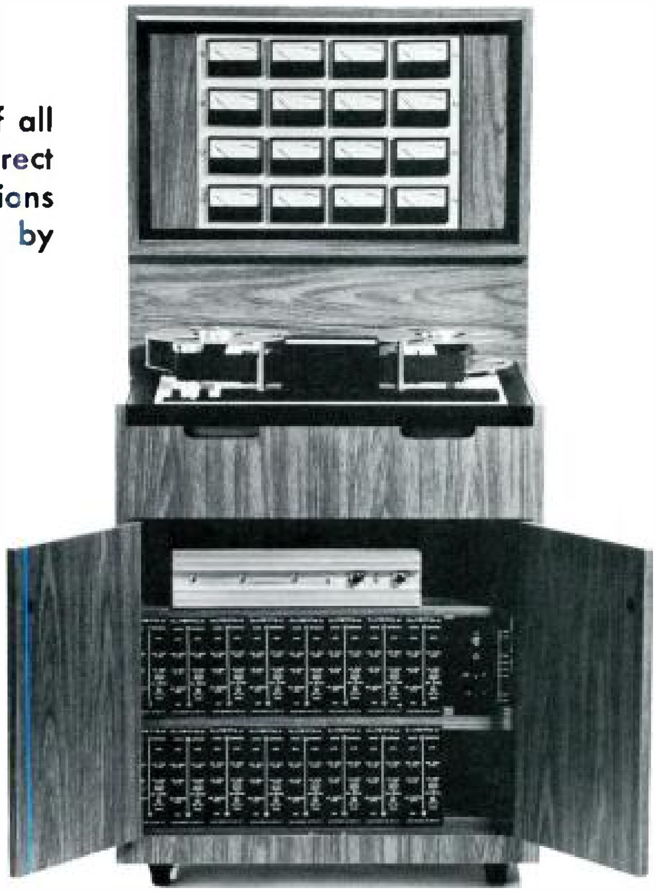
Next month, discussion shall center around circuits developed for short-distance, private two-way lines, long-distance telephone lines, and hybrid circuits. ■

# THE AUTO-TEC® RECORDERS L-16

The outstanding performance of the AuTo-Tec L-16 Recorder/Reproducer can only be compared with dual capstan, closed loop drive tape transports.



The simplicity of operation and design of all AuTo-Tec recorders/reproducers is the direct result of research and practical suggestions under actual studio working conditions, by experienced recording engineers.



COMPLETE WITH

Auto-Sense* Electronic Motion Sensing

"Auto Sync" Overdub Remote Control

"Instant-Change" Cue-Lift** Head Assemblies and Rotary Guides

*Pat. #3561700
**Reg./Pat. Pend.

# THIS IS IT!

**$19,995.00 COMPLETE**
F.O.B. NYC

*United Research Laboratory Corp.*

681 FIFTH AVENUE • NEW YORK, N. Y. 10022 • (212) 751-4663

*Circle 25 on Reader Service Card*

John M. Woram

# THE SYNC TRACK

• The literature on four channel continues to grow, and even if we don't get talked to death, as I suggested last month, there is a very real possibility of being buried alive. Some of the picturesque prose describing the joys of matrixed music is positively inspired. Did you know there is now available a phonograph cartridge, "... especially designed for derived four-channel stereo systems?" Just think about *that* for a while.

I suppose we shall all survive the claims and counter claims and eventually learn just what we can, and cannot, get out of our decoders. In the meantime, even if we now had the ultimate matrix—one that would *really* deliver four discreet channels of information, what should we expect to hear?

Dr. Ben Bauer of CBS Labs has pointed out that any sound source arriving from the rear will appear to be somewhat narrower than the same signal arriving from up front. To demonstrate this, listen to your stereo system in the usual manner. Then, turn around; the speakers will probably sound closer together than when you were facing them. In a four-channel system, perhaps the rear speakers could be spaced further apart than the front ones to compensate for this. But, regardless of where the rear speakers are placed, this *back-image contraction*—as Dr. Bauer calls it—suggests that whatever panning and directional devices we have been working with up front may not be as effective in the rear.

Once a sound originates behind us, how much localization facility do we lose? Probably quite a lot. In non-musical situations (which also seem to nicely describe a lot of the more recent four-channel spectaculars) we find ourselves turning to face any sound we are listening to. It's really rather difficult to follow a conversation going on behind us. I didn't say its impossible—certainly we can hear what's going on—it's just psychologically easier to pay attention when you're facing the source of sound. How many two-channel stereo systems have the speakers placed in the rear of the room? Few, if any. Obviously, people want to hear their music up front. If it didn't make any difference, a lot of decorating problems would be solved.

So, now we have four-channel sound, with rear speakers added to the system, and we should spend at least a little time thinking about the peculiarities of rear originating signals. Four channel infers a lot more than just two more speakers. Even in a discrete system, with four separate and equal signals, once we get past the speakers, the signals are no longer equal, due to a whole collection of factors, including psychoacoustics, construction of the (listener's) head, tradition, and lord knows what else.

A guitar to be localized in, say the left rear may have to be treated quite differently than if it was to be heard in the front. And then, when it is heard up front in a stereo program, it may not fit in as it should. I have the feeling that eventually producers are going to want to make separate stereo mixes, since musical four-to-two compatability problems are going to be a lot more severe than those of the old stereo-to-mono days.

The classical scene is, of course, another matter, though not without its own problems. Since with few exceptions the rear channels will consist of ambience, a satisfactory four-to-two arrangement can probably be worked out, although each of the contending matrix systems will probably present differing two-channel balances.

In the Columbia system, for example, rear-channel information should appear somewhat left and right of center when played back on a two-channel system. If this information is ambience, perhaps it might be more effective if it appeared at the extreme left and right front, and the actual front signal occupied the narrower off-center area, if this is indeed possible.

If the rear channels contain primary information, such as we might have in a Gabrielli double brass choir, it might be more advisable to arrange the four-channel program with the two choirs placed left and right instead of front and rear. That way, when played back in a two-track format, some directional feeling would be retained. This would be an example of compromising the four-channel format for the sake of a better two-channel playback.

In any case, it will probably be some time before we learn all the tricks of an effective four-channel recording technique, with or without matrices adding to the confusion.

On the subject of confusion, there's been a lot of talk lately about clockwise and counter clockwise helical grooves being cut on records for rear-channel information. It seems to me that one would find a helical groove on any record whenever one channel contains information 90 degrees out of phase with the other channel. Therefore helical groove cutting should not tax the abilities of any modern stereo cutter. Retrieving this helical information and routing it effectively to the rear is quite another matter, and is, of course, the function of the decoder. Although the ear is capable of focusing on a signal that leads another by 90 degrees, this still remains a marginally effective way of determining direction. It also brings up the question of how much deterioration of directional information will result from a slightly mis-aligned phono cartridge.

So far, I've managed to come up with a lot of questions, and few if any answers. The few answers each bring up more questions which only time, and a lot of experimentation, will eventually answer. And by then, we'll probably be just getting started with eight-channel systems. (*Let us hope not. Ed.*) ■

## classified

For more about the B&W 70CA speaker, write Linear Devices Inc., 148 French St., New Brunswick, N.J. 08901. Or call (201) 846.7777


Circle 20 on Reader Service Card

**Norman H. Crowhurst**

# THEORY AND PRACTICE

• Resuming the discussion of that 3-way crossover design problem, FIGURE 4 of the previous issue showed the crossover between mid-range and tweeter as using parallel-connected filters, while the one between woofer and these two used series-connected filters, to utilize the voice-coil inductance of the woofer.

This brings up two points that sometimes get asked about. In this particular case, would it be possible to use series connection also for mid-range to tweeter and thus utilize the voice-coil inductance of the mid-range unit? To which the answer is yes, quite probably: we will come back to that.

The other question relates to the relative merits of series *versus* parallel connection, in a more general way. And usually the questioner will follow up with some remark about parallel connection of loudspeakers being better than series connection.

We have discussed this series-parallel connection of speakers previously, particularly with reference to the effect that connection can have on damping factor, which is the main reason for preferring parallel connection: each speaker gets the low source resistance of the amplifier across its terminals, rather than having it in series with any other speaker(s) connected in series.

As has been shown before, this argument applies only when it is relevant. For example, where a number of similar units are mounted on the same baffle (a system that has advantages for wide-range use) so that the multiple area of the many speakers handle the bass, while each individual unit can handle higher frequencies than a single, large-diaphragm unit can.

In this case, because all the units are feeding the same air load, they share damping problems, and thus the damping factor of the whole assembly is what matters, so that series connection, parallel connection or series-parallel connection are equally suited provided the connection results in a match for the amplifier.

However, the case that started this question is quite different from that. Those discussions of series versus parallel all involved the use of speaker units that handle the whole frequency range, whether they are acoustically linked or not. This involves the use of units that cover different frequency ranges and in which the crossover filters serve the function of separating the frequencies fed to each.

Because of this, the speaker units connected to the various outputs of the crossover are not in series or parallel with units connected to other outputs in the usual sense. The amplifier's damping factor gets applied to each unit directly, over its relevant frequency range, and is uncoupled from it as the termination of the range is approached.

So the better, as well as the more economic, method of selecting an appropriate crossover network, is to consider the nature of the speaker impedance the networks feed. As the woofer's voice-coil inductance can be used as part of its low-pass filter, so can the mid-range be considered.

However this should not be taken as a rule. It may be that the rise in reactance due to voice-coil inductance occurs at a point on the frequency range where the speaker begins to lose its value as a transducer, due to mechanical or acoustic effects, or it may not.

Testing whether it can or not may follow the same procedure used to check out the woofer's impedance characteristic, described in the previous issue. If the inductance value does cause a reactance rise in the crossover region, this may fix frequency for you. If not, then you have the choice of crossovers (FIGURE 1).

In this event the choice may depend on availability, or suitability of values. For a specific crossover frequency and impedance of operation, the reactance values are just double, in the parallel-input circuit, what they are in the series-input circuit.

Suppose we settle on 2500 Hz as the mid-range/tweeter crossover, for whatever reason, and the impedance, as already settled, is 16 ohms. For the parallel-input arrangement, each reactance needs to be 1.414 times 16 ohms, or 22.6 ohms, at 2500 Hz. For the series-input arrangement, the reactances need to be 0.707 times 16 ohms, or 11.3 ohms, at the same frequency.

At 2500 Hz, omega is $6.28 \times 2500 = 1.57 \times 10^4$. Substituting this in the reactance formula gives the capacitor values as 2.8 mFd for the parallel-input circuit, or 5.6 mFd for the series-input circuit. Similarly, for inductances, the choice of values is 1.5 millihenries for parallel-input or 750 microhenries for series-input.

The question as to whether the voice-coil impedance influences the choice or not is not a hard-line answer. In this instance, if the voice-coil has an inductive component of 750 microhenries, which produces a reactance of 11.3 ohms at 2500 Hz, it is just right to do without the output inductor to the mid-range unit. But what if the voice inductance computes to 500 microhenries, for example?

You have two choices: if the 2500 Hz crossover was critical, due to reproduction considerations, such as that the mid-range units response becomes irregular above this frequency, then you could add a 250 microhenry inductor as an external component

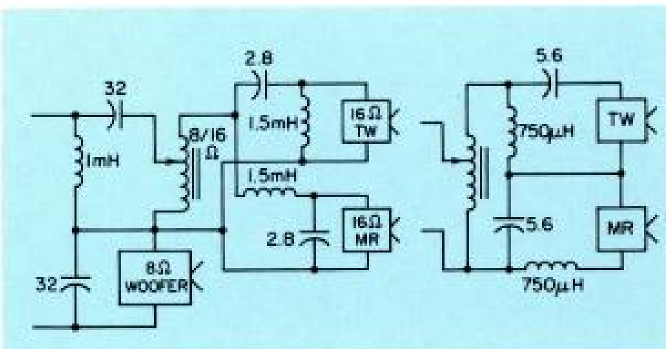



*Figure 1. Alternative ways of connecting a crossover to the 16-ohm side of the impedance conversion.*

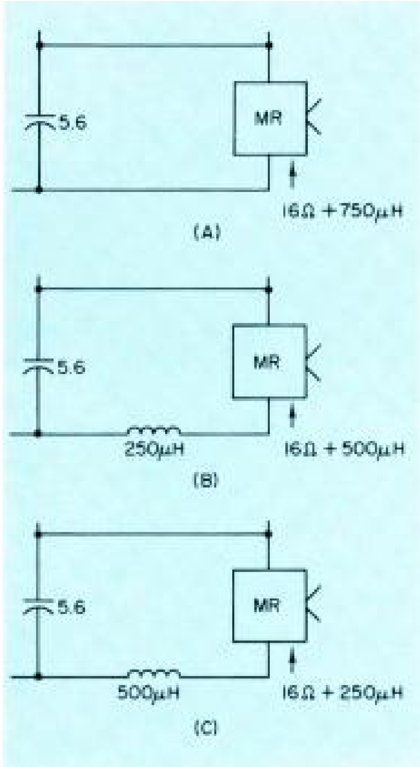



*Figure 2. Possible compensations for voice-coil inductance of the mid-range unit: (a) where voice coil provides the whole 750 microhenries; (b) where voice coil is 500 microhenries; (c) where it is 250 microhenries.*

(FIGURE 2); on the other hand, if frequency is not critical for such reasons, you may prefer to shift your crossover frequency. But check the reactance at the new frequency. If inductance is constant, the reactance should be 11.3 ohms at 1.5 times 2500 Hz, which is 3750 Hz. Having verified this, and shifted your choice of frequency if necessary, you can complete the design.

But what if the inductance of the voice coil is only about 250 microhenries, producing a reactance of about 3.8 ohms at 2500 Hz, and you definitely do not want to put your crossover frequency up to 7500 Hz. Should you use a 500 microhenry inductance, or is it satisfactory to go to the other configuration (if the values happen to be more convenient)?

The question here really rests on whether ignoring the reactance component of the voice-coil impedance will seriously affect performance. Crossovers are designed on the theoretical basis of each output being terminated with a resistive impedance of the prescribed value. Loudspeakers seldom have a purely resistive impedance, except at a few odd frequencies, and the value of impedance varies as the phase swings back and forth with changing frequency.

Before you start to conclude that it's a wonder crossovers work at all, we should point out that values only become critical at frequencies close to crossover. At frequencies well within the pass band of the filter, the network simply transfers whatever impedance is connected to its output, back to its input.

A good way to think of any kind of crossover network is as a frequency-operated switch: below crossover frequency, the impedance of the woofer (or mid-range, as the case may be) is connected to the amplifier output; above crossover frequency, the impedance of the mid-range (or tweeter) unit is connected to the amplifier output.

But the switch is not quick-acting: it does not happen suddenly, at the crossover frequency, but gradually, as frequency passes through the crossover value. And because the switch is made of reactance elements, reactances in the impedances being switched can influence its action, at these frequencies only.

So there is no way to eliminate the effect of reactances in the vicinity of crossover. If they are small enough—say less than one-third the working impedance—maybe you can ignore their effect, for all intents and purposes. If they are larger, you must take them into account, or find the crossover does not do what you expect it to do. ■

## you write it

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

Are you doing something original or unusual in your work? Your fellow audio pros might want to know about it. (It's easy to tell your story in **db**.)

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

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

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

And at $16,500, the JH-16 is worth fighting over.

Combining the total logic and the constant tape tension of the JH-10 Transport with proven MCI electronics, the JH-16 is a triumph of both lower cost *and* higher quality.

MCI has eliminated the costly hand-wiring found in other comparable units and substituted channel strip printed circuit boards in conjunction with non-redundant functions. Yet, there's no sacrifice of world-renowned MCI quality—in fact, serviceability and reliability are improved.

The JH-16 is a complete 3-head 16-track recorder that can be changed in minutes for one- or two-inch tape capability, for 8-, 12- or 16-track operation.

Remote overdub and transport motion control are standard equipment. Space is available in the attractive aluminum housing of the JH-16 remote control for the optional Auto-Locator ($1,200).

Atlantic Records' Tommy Dowd and Criteria Recording Studio's Mack Emerman aren't the only ones fighting for the first JH-16. To join the fray, contact MCI, 1140 N. Flagler Drive, Ft. Lauderdale, Fla. 33304 (phone 305/763-5433).

# The big problem that MCI has with its JH-16 is:

# WHO'LL GET THE FIRST ONE?

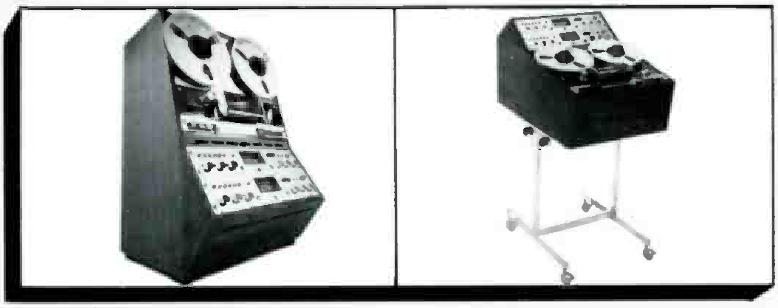
MCI DISTRIBUTORS: Tom Hidley, Westlake Audio, 6311 Wilshire Blvd., Los Angeles, Calif. 90048 • Dan Fleckinger, Dan Fleckinger Associates, P.O. Box 628, Hudson, Ohio 44236 • Dave Harrison, Studio Supply Co., 112 Cloverdale Court, Hendersonville, Tenn. 37075 • Paul Kelly, Kelly's Audio Engineers, 704 Elmhurst, Muscle Shoals, Ala. 35660.

MCI

*Circle 29 on Reader Service Card*

# ANY WAY YOU LOOK AT IT OR ANY WAY YOU USE IT YOU CAN'T BEAT IT!


OTARI
MX 7000



See the dynamic
4-channel MX 7000QX!

**THE OTARI MX 7000 PROFESSIONAL TAPE DECK** is designed with you, the professional, in mind. Built with the same rugged dependability of the OTARI 32:1 Duplicator and Q.C. Monitors, this outstanding unit has these exclusive features:

- **Three speed hysteresis synchronous motor (3¾, 7½ and 15 IPS). (Adapter kit available for 30 IPS).**
- **Convenient built-in test oscillator at 700 Hz and 10 KHz.**
- Plug-in head assemblies for changing from one to four channels.
- Push button controls
- Sound with sound; sound on sound.
- Automatic equalization in all 3 speeds.
- 10½" reel capacity
- Can be used in vertical or horizontal position by merely reversing the modules within cabinet.
- Prices start at under $2000.00

*For more details, specifications or a representative, Phone, write, TWX or wire*

**OTARI**

**OTARI OF AMERICA, LTD.**
8295 South La Cienega Blvd., Inglewood, Cal. 90301
Telephone: (213) 678-1442 ■ TWX 910-328-6184

OTARI ELECTRIC CO. LTD., SUGINAMIKU, TOKYO, JAPAN

*Circle 26 on Reader Service Card*

# NEW PRODUCTS AND SERVICES

## EXPANDER-NOISE REDUCTION SYSTEM


• The model 117 restores lost dynamics to records, tapes, f.m. broadcasts. Decilinear (linear decibel) expansion is achieved with true r.m.s. sensing and wide dynamic range analog computer gain control circuits. Used as a noise reduction system it can improve tape recorder signal to noise ratio by 20 dB. Compatible with any high quality component music system. Specifications include: 100 dB dynamic range; compression - expansion continuously adjustable from 0.5 slope (10 dB output change/20 dB input change) to 2.0 slope (20 dB output change/10 dB input change); distortion less than 0.5 per cent at 20 Hz for complete compress-expand cycle.
*Mfr: dbx, Inc.*
*Price: $145*
*Circle 56 on Reader Service Card*

## CONDENSER MIC

• Model C-412 is an f.e.t. condenser microphone system using a one-inch diaphragm capsule. The unit incorporates a pattern selector switch to vary polar response from omni to cardioid to figure-8. In addition it is equipped with a 20 dB switchable output pad. It may be phantom powered by utilizing a 24 V B+ supply or with a.c. or d.c. supplies available from the company.
*Mfr: AKG*
*Circle 50 on Reader Service Card*

## SPECTRUM ANALYZER

• A real-time audio spectrum analyzer has been designed to provide high quality at what is claimed to be a price breakthrough. The 8050A is a compact analyzer that covers a frequency range of 40 Hz to 16 kHz with 27 parallel bandpass filters. Detectors following the filters convert the a.c. filter outputs to d.c. levels proportional to the r.m.s. value of the a.c. signals. An internal scanner sequentially connect the 27 detector outputs to the c.r.t. display screen through a log converter and simultaneously generates a linear ramp at the *X* deflection. Preamp gain is variable from −20 to +20 dB. The unit contains a built-in power supply to operate condenser-type microphones.

*Mfr: Altec Division*
*Price: $3018*
*Circle 51 on Reader Service Card*

## TAPE RECORDER

• A new entry in the American market comes from Germany with the model TG 100 stereo tape deck. It features many interesting design techniques including inaudible drive noise, a photo-electric control of tape tension for both reels which makes the tape tension on either side of the capstan almost independent of the reel diameter. All metal mirror heads are used with provisions for a fourth head in its plug-in head assembly. This may be used for 4 track playback or 4 channel playback or used for slide synchronizing. Unit includes a built in mixer, sound on sound, echo and tape monitoring. The tape drive is extremely gentle so that even triple play tape can be used. Peak reading level meters are used for fast response. All functions are relay controlled and solenoid actuated. An optional remote control is available. Three motors are used with an electronically controlled d.c. motor for capstan drive. Frequency response is 20 Hz to 25 kHz at 7.5 in/sec. and the machine is supplied with three-speed operation of 7.5, 3¾ and 1⅞ in/sec.
*Mfr: Braun of Germany*
*Price: $689.50*
*Circle 69 on Reader Service Card*

## QUADRAPHONIC MIXER

• Sixteen inputs are positioned in 4 quadrants by concentric pan pots. These new pots are identical circuits to the joystick variety. However, they offer the added feature of compactness, low cost and the concentric knobs can be calibrated to allow easy return to logged positions. The two joystick pan pots increase the input capability to 18. This provides for final mixdown additions of tape echo or special solo tracks. Four echo return pots are provided, plus four balance pots and master gain control. The basic design allows conventional stereo mixdowns to be accomplished without modifying the quad panner connections to your present console. The MQ 184 mini quad panner system is a unity gain device intended to operate at a nominal + 4 dBm in and out. The signal-to-noise ratio is 70 dB. All input impedances are 600 ohms unbalanced, including the 4 echo returns. The four outputs are 600 ohms, transformer isolated. Power requirements are 115 or 230 VAC, 50 or 60 Hz.
*Mfr: Quad-Eight*
*Price: $2795*
*Circle 67 on Reader Service Card*

## 'SCOPE CAMERA

● A low-cost camera for producing instant hard copies of oscilloscope displays in just 15 seconds has been introduced. Unlike most permanently-mounted cameras, a single hand-held CR-9 can easily be used with more than one oscilloscope at multiple installations. Because of its portability and light weight, the unit can easily be carried from one terminal to another. The CR-9 is designed to accept any of 8 interchangeable, light-tight hoods in round, square, and rectangular configurations which fit almost all oscilloscopes with 6 x 8, 6 x 10 and 8 x 10 cm graticules. The hoods quickly attach to the front of the camera, permitting only light from the cathode ray screen to reach the film. Additionally, the hoods position the camera at the correct distance from the display, assuring sharp, virtually distortion-free pictures. Object-to-image ratio is 1:0.85, which permits recording an entire 8 x 10 cm graticule on the 3¼ x 4¼-inch Polaroid instant print. To make instant hard copies, the CR-9 camera is placed against the cathode ray screen and the trigger squeezed to release the shutter. Since film development takes place outside the camera, the operator is free to take pictures in rapid sequence. Shutter speed and lens opening vary depending on the brightness of the display and the type of phosphor used. Once initial camera settings are determined for a given test series, exposure changes are seldom required. The camera is designed to use standard Polaroid Type 107 pack film which drops into the back of the camera. Each film pack produces eight 3¼ x 4¼-inch pictures. Development time is only 15 seconds.

*Mfr: Polaroid Corp.*
*Price: $179.95*
*Circle 61 on Reader Service Card*

## MIXER-AMPLIFIER

● This solid-state mixer/amplifier has been designed to meet the performance requirements of the professional user. It is equipped with five input channels each of which adapts to microphone, line or magnetic phono at the turn of a switch. Each channel is furnished with three levels of padding (−10, −20, and −30 dB) for use in preventing overloading of the system by loud voices or music. Rated at +28 dBm transformer isolated output, model 2A75 has ample headroom (gain) to handle peaks when operated at +18 dBm. Pleasing in appearance and highly practical, this unit is supplied with an illuminated professional quality vu meter, controls whose positions can be determined by sight and touch, and in attractive light gray, baked enamel finish. The unit also features a second output channel permitting premonitoring of the program before and during distribution to the main output channel. It is available for mounting in a standard 19-inch rack or in a portable housing. Specifications include a frequency response of 20 to 20,000 Hz ± 1 dB, distortion of less than 0.5 per cent at +28 dBm, 20 to 20,000 Hz, and a power requirement of 105-125 volts, 50-60 Hz, 10 watts, or +25 volts d.c. at 100 mA.

*Mfr:* DuKane Corp.
*Circle 62 on Reader Service Card*

## SEMICONDUCTOR TESTER

● The TT-7 is for testing transistors and diodes and troubleshooting solid-state equipment. Provides dynamic go/no go indication of semiconductor status in circuit or out. No clip leads to attach and no meter to watch thus providing single handed, eyes on work testing. Special probe conforms to all transistor configurations and styles without adjustments. The TT-7 also enables the user to determine type of transistor (NPN or PNP) and unscrambles lead configuration on unmarked units. Although the unit is simple in nature it has proven itself to be an effective instrument for troubleshooting. It is completely safe for even the most delicate semiconductors no matter how incorrectly applied. Operates on two inexpensive penlight cells and will operate for months without being turned off.

*Mfr: Ramko Research*
*Price: $16.95*
*Circle 64 on Reader Service Card*

## AUDIO GENERATOR

● A solid-state audio sine-wave generator featuring low hum and distortion throughout the life of the equipment is now available. The model F380A is intended for manufacturers and users of professional audio and broadcast equipment. The generator covers the 20 Hz to 20 Hz frequency range in three continuously variable decades. Output is +15 dBm into 50, 150 and 600 ohm impedances; the output is transformer coupled, permitting either balanced or unbalanced configurations. Hum level is more than 75 dB below the output voltage, or more than 100 dB below zero level.

The output attenuator provides up to 111 dB in 0.1 dB steps. A voltmeter continuously monitors the output in volts or dBm. Frequency response is within ±0.2 db from 30 Hz to 15 kHz. The instrument makes use of a Wein bridge to limit distortion to less than 0.25 per cent from 20 Hz to 100 Hz, and less than 0.1 per cent from 100 Hz to 20 kHz. The distortion is normally not affected by component aging, unlike earlier tube models, which tended to increase in distortion (as well as noise and hum) with age.

*Mfr: Microdot, Inc.*
*Price: $925*
*Circle 59 on Reader Service Card*

## STEREO MIC MIXER

• A flexible stereo microphone mixer, designed to provide excellent versatility for the serious audiophile as well as the professional sound engineer, has been introduced as the model M688. It will deliver high quality performance—particularly when used with stereo tape recorders without built-in mixing capability. It is also ideal in audio-visual and multi-media presentations in which a stereo music source is used, or for stereo sound reinforcement applications. The professional user will find the M688 useful for stereo recording of musical or dramatic events, while simultaneously providing a left plus right (mono) microphone level output for use in the sound reinforcement system for the presentation. It accepts four high or low impedance microphones through four inputs, plus a stereo auxiliary high level input, each with its own volume control. Three microphone inputs have front panel switches for left or right channel output. The fourth microphone input has a pan control which allows this input to be directed to the left channel, the right channel, or anywhere in between. A stereo master volume control simultaneously adjusts the level of all inputs. If further inputs are needed, the unit can be paralleled with M67 mixer or another M688 via mix bus jacks. The mixer has a stereo high level high impedance output plus a left plus right (mono) microphone level output (Hi or Lo Z).

*Mfr: Shure Bros., Inc.*
*Price: $114.00*
*Circle 60 on Reader Service Card*

## BROADCAST CONSOLE

• The AS-40B stereo console has 8 stereo mixing channels allowing 14 audio source inputs plus an auxiliary 3 station input push-button bank. The unit incorporates left and right program amplifiers in the audition and program outputs, in addition, the operator can monitor audition and program sources by means of a selector switch under the VU meters.

The AS-40B features a pair of "balance" controls on the front panel of the console in conjunction with the program and audition ouputs. This allows the operator to quickly compensate for any erratic left to right channel imbalance without changing the master gain controls.

Left and right microphone preamplifiers are provided in channels 1 and 2. An optional left and right channel microphone preamplifier may be installed in channel 3. Mixing channel 1 is designed as the main microphone source with 3 selectable inputs in addition to a special stereo/mono mode selector switch and microphone pan pot.

All mixing channels have a cue position and self-contained cue amplifier. The head phone station has a separate gain control.

The basic Model AS-40B Stereo Console includes external power supply. The Model AS-40B-1 includes the MAS-50 50 Watt Stereo Power Monitor Amplifier with plug-in SM-3 muting relay accessory.

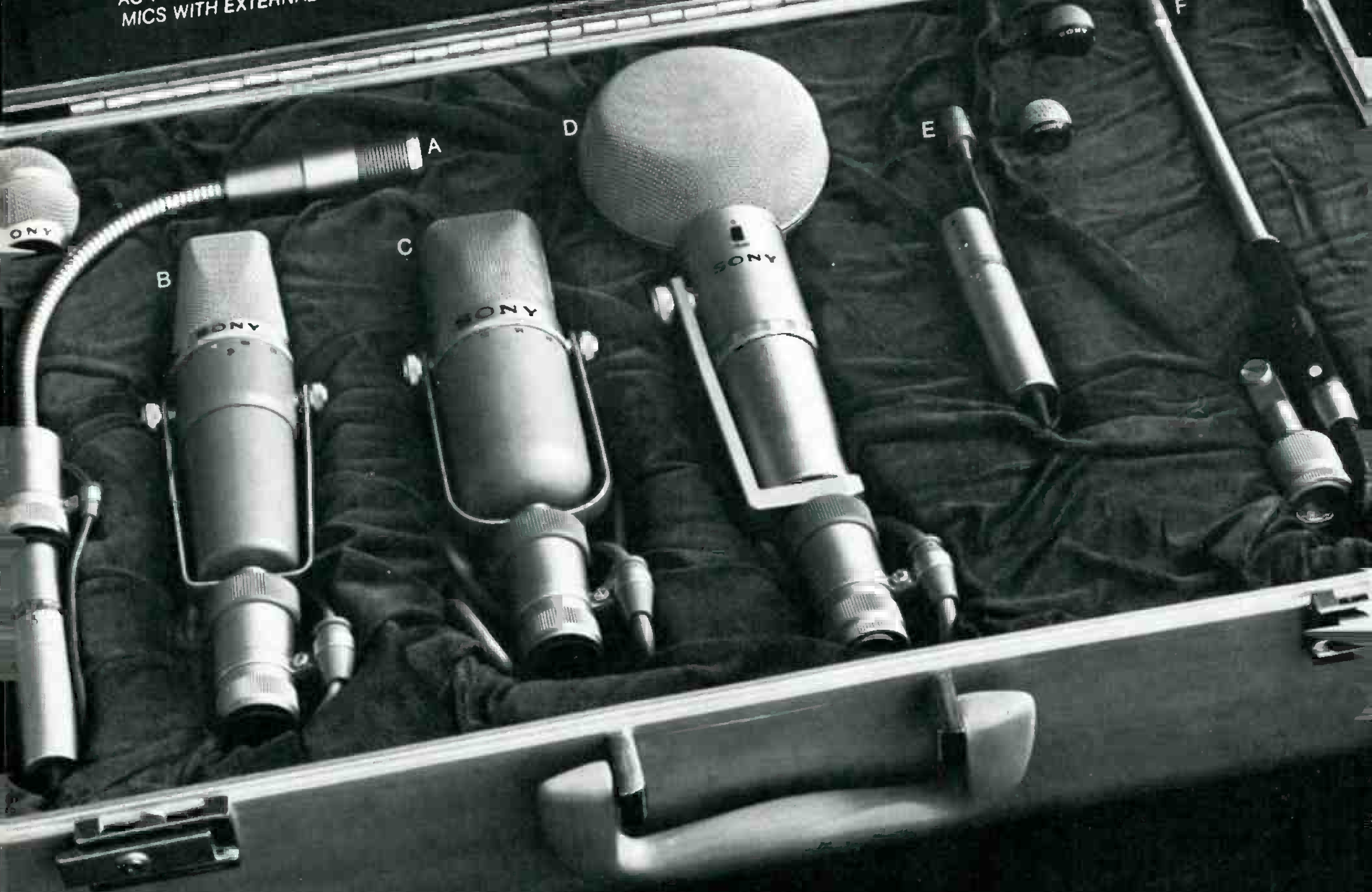
*Mfr: Sparta Electronic Corp.*
*Price: AS-40B—$1770; AS-40B-1—$1995*
*Circle 54 on Reader Service Card*

## RADIAL HORN

• This weatherproof horn is specifically designed for high intensity audio in the middle and upper range of the spectrum and features a controlled, resonance free, directional dispersion pattern. It is recommended for application in paging and public address systems and can be used either as a single projector horn, in a horn array, or as a high frequency component of professionally designed, full-range loudspeaker systems. The RH-500 is a straight radial horn with controlled dispersion angles of 120 degrees horizontal and 40 degrees vertical. It has a 500 Hz. low cut-off frequency. The horn is manufactured of one-piece cast aluminum. It is

equipped with a coupler throat which will accept standard 1⅜ inch threaded, high compression driver units. Size is 27¾ inches wide, 8½ inches high and 18⅜ inches long.

*Mfr: Atlas Sound*
*Circle 71 on Reader Service Card*



A
B
C
D
E
F
SONY

MARSHALL KING

# The Sync Pulse in t.v. Audio, Part 3

*Picking up where we left off last month, the author establishes firm guidelines for the synchronizing of video to audio recorders.*

***It is hoped . . . that some standardization of method is not far away.***

NOW THAT the groundwork was laid last month with regard to definitions and aims, let's hazard to state some of those "infallible guides" we're looking for, then see if the analyses which follow will substantiate their worth. These seven "rules" are nothing more than what was left over after discarding one bad practice after another in many years of doing it the hard way.

*Rule 1. Playback will be synchronous if the ratio of sync pulses per second between the two carriers during recording is the same during playback.* This seems too simple to be true, yet herein lies everything that is said in the other six rules which follow. Such synchronous playback can be brought about through either *autonomous* operation or *slave* operation.

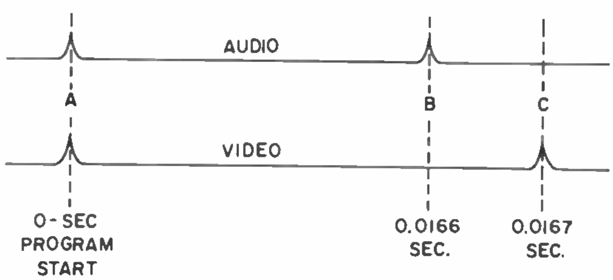
*Autonomous* operation is where, after starting together during playback, the reference marks on both a carrier and its companion refer to the constant frequencies from which they were derived. *Autonomous* operation is possible if—

*Rule 2:* during recording, the sync on each carrier, which may be of a different frequency from that of its companion, comes from a generator of constant frequency, and—

*Rule 3:* during playback, both a machine and its companion have a means of reaching a common start mark at their own operating speeds, and—

*Rule 4:* during playback, both a machine and its companion have servo monitoring between carrier speed and the generator from which sync was derived during recording.

Figure 1. A method of autonomous operation.

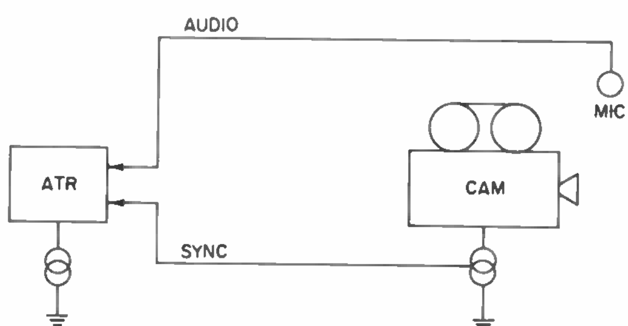



Suppose we have reason to record an orchestra on two audio tape machines so that we may switch from one tape to the other during simultaneous playback. And for the sake of example only, let's say that during recording a constant frequency of 10,000 Hz was put on the tape of machine A and 50 Hz on machine B. The ratio of sync A to sync B is 200. If this ratio is maintained during playback, the two machines will stay in sync simply because the sound on the two tapes will be going by their respective playback heads at the same rate. This would happen if both machines played back in the same time it took to record. It would also happen if, for some unlikely reason during playback, the sync pulses on machine A passed the tach head at a rate of, say, 12,600 Hz and those of machine B at 63 Hz. The ratio here is still 200, and because of this, synchronous operation will prevail, even though the sound will not be a true pitch.

But suppose, during playback of these two machines, the ratio became, say, 201, or perhaps 196. Since the *record* ratio of 200 is not being maintained, it is evident the two sound tracks are not going by their respective playback heads at the same rate. It is evident, therefore, that when the sync-pulse ratio changes between two machines from record to playback, synchronous operation is impossible. This is the basic statement of *Rule 1.*

A more practical example is where the video tape has recorded a steady 59.95 Hz on its control track and the

Figure 2. Although it may not look this way, this is a slave operation.



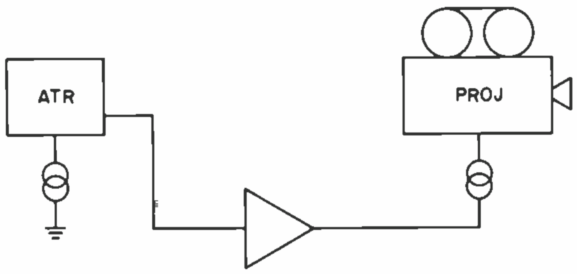



*Figure 3. Since both carriers contain corresponding pulses, it is possible for the reproducer to drive the projector.*

audio tape has a steady 60 Hz on its sync track. If both machines were to resolve to (or take their playback speed-control instructions from) either of these frequencies alone, they would soon be far out of step. In a 90-minute program they would be 4½ seconds out of sync at the end of the program if they started together. But if the v.t.r. resolves to its 59.95 Hz and the a.t.r. to 60 Hz, the picture and sound will pass their respective playback heads at the same rate, and thus stay in step.

A close analysis of this shows that for every sync pulse that passes the tachometer head on the audio tape machine, there is a certain amount of audio (as measured in real recording time) reproduced from the playback head; and for every sync pulse that passes the tachometer head on the video tape machine, a *different* amount of video is reproduced. But this does not mean the two are out of sync, for the ratio of these two amounts of program (as measured in real recording time) is the same as the ratio of their two sync frequencies.

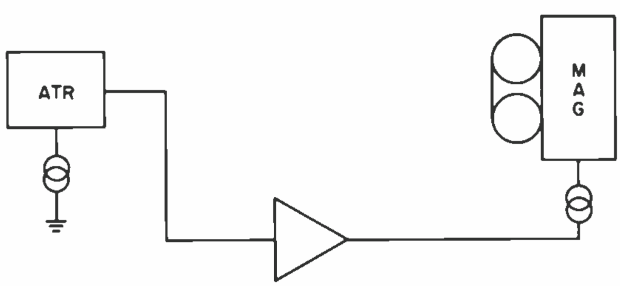
FIGURE 1 shows this facet of *autonomous* operation quite well. Approximately 0.0166 seconds (= 1/60 sec.) transpires between audio sync pulses, and approximately 0.0167 (= 1/59.95 sec.) between video sync pulses. But the sound that plays back between adjacent audio sync pulses (points A and B) is accompanied by a simultaneous playback of a corresponding segment of picture, which does not require all the distance between corresponding adjacent *video* sync pulses. By the time point C has been reached, the audio has passed its second sync pulse and the video has just arrived at *its* second sync pulse, and all the while the picture and sound have been in step. In other words, for both picture and sound, a limb breaks from a tree at point B and hits the ground at point C, regardless where the sync pulses are.

Thus, we have met the parameters for *autonomous* operation, and they coincide with the assumptions of both primary *Rule 1* as well as *Rules 2, 3,* and *4* for *autonomous* operation.

*Slave* operation is where, during playback, the reference marks on one carrier passing its tachometer head causes corresponding marks on its companion to pass its tachometer head at the same time. *Slave* operation is possible if—

*Rule 5:* during recording, the sync on each carrier, whether constant or not, comes from the same source, and—

*Figure 4. This slave transfer is desirable.*



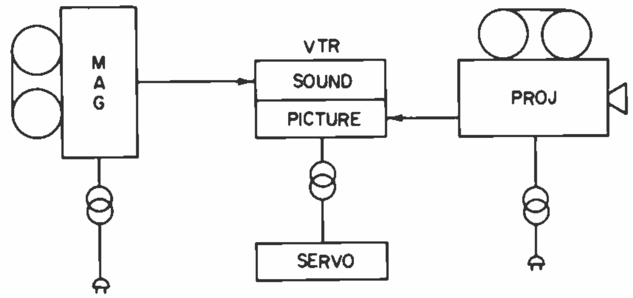




*Figure 5. The synchronous action of the transfer described in the text.*

*Rule 6:* during playback, the synchronous motor of the *slave* machine is driven by the output from the sync track of its companion, and—

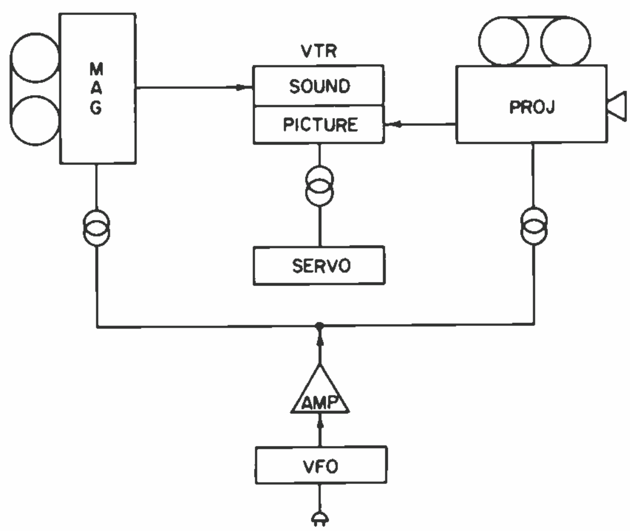
*Rule 7:* during playback, the *slave* machine has servo monitoring between its carrier speed and the incoming reference sync from its companion.

From this it can be seen that we cannot play back, under *slave* operation, the two machines in the above example, since the sync on each carrier did not come from the same source: one came from 60 Hz and the other from 59.95 Hz. In the 90-minute referred to, there are a total of 323,730 pulses laid down on video tape and 324,000 on audio tape. If we were to slave the video-tape motor to the audio-tape sync track so they played back pulse for pulse, we can see what would happen. By the time 323,730 pulses had gone by the picture would have come to an end, and audio would still have 270 pulses to go. At the rate of 60 Hz audio would be running 4½ seconds after the picture ended.

But let's say that both audio and video machines recorded a sync pulse of 59.95 Hz on their tapes, as shown in FIGURE 10(A). Under methods already discussed, they could play back together in *Autonomous* operation, or, as indicated in FIGURE 10(B), they can play back (or transfer) in *slave* operation.

Here the video tape is a slave to the audio tape. For every sync pulse on the audio tape that goes by its tach head, a certain amount of program material—the amount existing between sync pulses—goes by the playback head. Also, the transmission of this pulse to the v.t.r. machine causes its motor to turn one revolution. This, in turn, moves the video tape an equivalent distance, for that was the ratio of sync pulse to rpm during recording. To pre-

*Figure 6. The reproducer and projector both driven by 60-Hz line frequency. But they may not playback in the same time as it took to record them.*



| | RECORD | | TRANSFER | | PLAYBACK OFF LINE | | PLAYBACK OFF VFO | |
|---|---|---|---|---|---|---|---|---|
| | CAM | ATR | ATR | MAG | MAG | PROJ | MAG | PROJ |
| CONSTANTS: TOTAL PULSES | 3000 | | | | | | | |
| CONSTANTS: TOTAL FRAMES | 1200 | | | | | | | |
| CONSTANTS: PULSES PER FRAME | 2 1/2 | | | | | | | |
| VARIABLES: FRAMES PER SECOND | 20 | – | – | 16 | 24 | 24 | 20 | 20 |
| VARIABLES: TOTAL TIME | 60 SEC | 60 SEC | 75 SEC | 75 SEC | 50 SEC | 50 SEC | 60 SEC | 60 SEC |
| VARIABLES: PULSES PER SECOND | 50 | 50 | 40 | 40 | 60 | 60 | 50 | 50 |
| VARIABLES: INCHES PER SECOND | – | 15 | 12 | – | – | – | – | – |

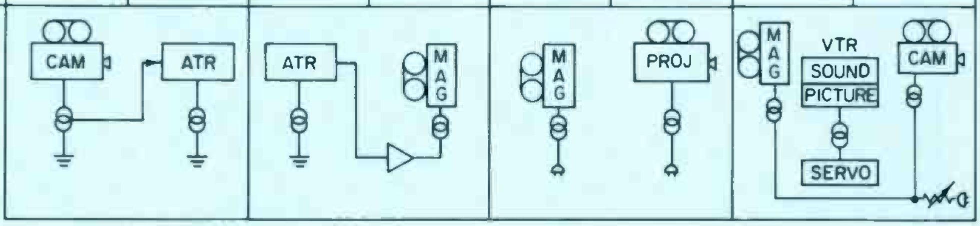



*Figure 7. A chart showing how possible discrepancies came about.*

clude the destruction of this ratio due to slippage during playback, the servo circuit in the v.t.r. machine monitors the relationship very carefully.

Although the video and audio machines will stay in sync this way, there won't be a usable picture if the v.t.r. playback speed varies more than a very slight amount from 59.95 Hz. Therefore, an intermediate *slave transfer* of audio to a clean video tape is advised, as explained earlier in Part 1 and shown here in FIGURE 10(B).

A more common case where *slave* operation is used is where the picture is being recorded on film in a battery-driven camera and the sound on tape in a battery-driven recorder. This situation is almost mandatory in locations where a.-c. line power is not available, and although it's a technique originally applicable to motion pictures, the resultant product often makes its way to video tape through transfer.

It is interesting to note that although the recording set-up as shown in FIGURE 2 seems to be neither *Autonomous* nor *slave*, it is actually *slave*, for each machine is receiving sync from a common source, the sync source is not necessarily stable, and the sync pulses are being laid down on each tape (or film) at corresponding points in the program as will be shown.

The reason that the sync frequency may not be stable is that they are being generated by a battery-driven motor/generator, and batteries do run down. Let's say that during recording the camera runs at a speed below normal, and during transfer to magstripe the reproducer runs slower than normal, and that during playback both machines run at normal line frequency.

The camera motor generates one pulse for every revolution and this is fed to the sync track of the audio recorder. Since in the United States it has been established that a normal line frequency shall be 60 Hz and that a normal frame rate in film cameras shall be 24 frames per second, we can calculate that for 60 Hz to move the film at 24 F/sec, the film must move one frame for every 2½ turns of the camera motor, and it is designed to do so.

Since in *slave* playback it is desired (by definition) that one pulse from the prime carrier causes one pulse of travel in the *slave* carrier, it is important that, during recording, on each carrier a pulse is laid down from the same source, so that unlike *autonomous* operation, sync pulses on both sound and picture carriers are required to fall at corresponding points in the program.

The question now is how to record a sync pulse on the picture carrier, or film. It is done by the manufacturer by the simple expedient of putting sprocket holes in it. These will of course serve as our sync pulses provided there is a constant relationship between the pulse generated by the motor/generator and the number of holes moved by each pulse (which means by each turn of the motor). Such indeed is the case, for if there are four sprocket holes per frame in the film and 2½ pulses per frame, then there is a constant ratio of one pulse for every 1.6 sprocket holes. In effect we can say that for every pulse laid on audio tape there are 1.6 holes punched in the film. Or, if we use a conversion factor where 1.6 holes equals one film pulse, we can say that one audio pulse equals one film pulse.

It is important to note here that since the sync pulses on both carriers come from the same source, they fall (in each carrier) at corresponding points in the program, which is a requirement implied in the definition of *slave* operation.

Since both carriers, after recording, contain corresponding sync pulses, we now have a means of synchronous playback. It is possible now for the reproducer to drive the projector, as shown in FIGURE 3. Regardless of any speed fluctuations in the audio machine, the film projector will follow faithfully, one pulse for one pulse. (Or to be more precise, one pulse for 1.6 sprocket holes. It's important to note that it's not how many sprocket holes per inch or per frame that concerns us, so much as it is their consistent relationship to the electronic sync pulses. Without this consistency we would have no conversion between the two and thus could not regard the holes as pulses.)

The above synchronous playback will work, but it is often more desirable to first transfer the sound to a sprocketed magnetic film (magstripe) so that an unlimited

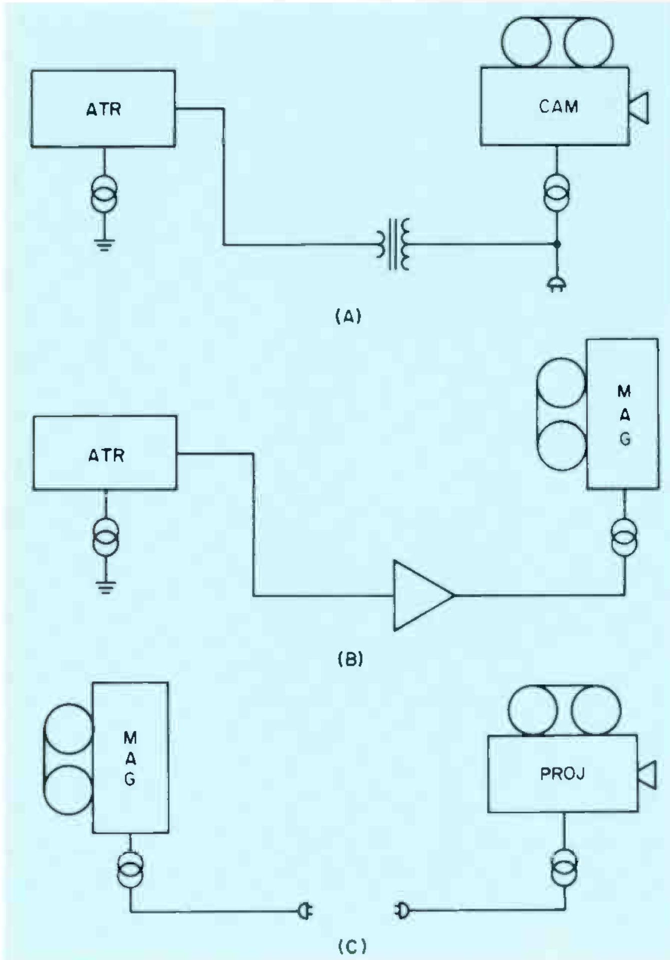



Figure 8. Ways of obtaining synchronous action. (A) either; (B) slave; (C) auto.

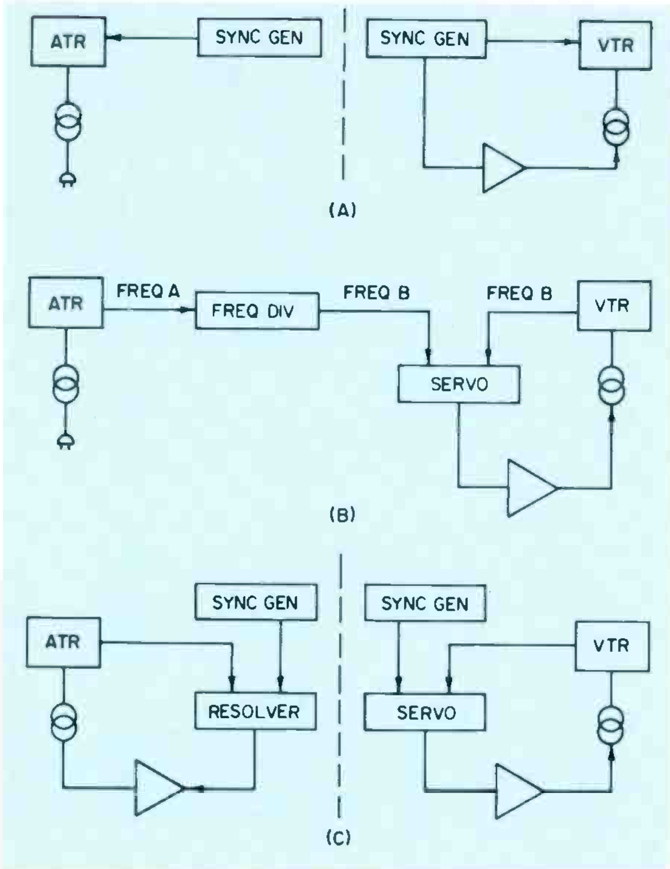



*Figure 9. As in Figure 8, again, (A) is either; (B) is slave; (C) is auto.*

number of additional sound tracks (music and effects) can be combined in sync with our original program audio. So the *slave* transfer shown in FIGURE 4 is usually the next order of business. There is no picture involved here but its presence is implied. It "exists" on the magstripe only in phantom, in that for every four holes (2½ sync pulses) of sound we lay down on the magstripe, there is a corresponding frame of picture on another reel somewhere that will be locked up to this magstripe when this audio transfer is finished. The synchronous action in the transfer is precisely as described for the projector playback of FIGURE 5.

After *transfer* the operation becomes *autonomous*, for both playback machines—magstripe reproducer and film projector—are locked together through sprocket holes and gears and are referred to a common frequency for motor drive, rather than to each other directly. We are through with sync pulses. It remains now to transfer the combined output of these synchronous machines to video tape, which is *autonomous* in subsequent handling of the product.

But there is something wrong here. If the reproducer and projector are driven at the line frequency of 60 Hz as shown in FIGURE 6, the program they are reproducing will not play back in the same time it took to record it, under the circumstances we have set up. If during recording the camera ran at 20 f/sec instead of the normal 24 f/sec, and if during transfer the reproducer ran at 12 in/sec instead of its normal 15 in/sec, then a 60-second commercial will play back in 50 seconds. This will never do for those who are paying the bill, and an adjustment must be made by running the reproducer and projector off 50 Hz instead of 60 Hz. An analysis of this situation might be worthwhile, and the chart in FIGURE 7 shows how the discrepancy came about and how it is handled

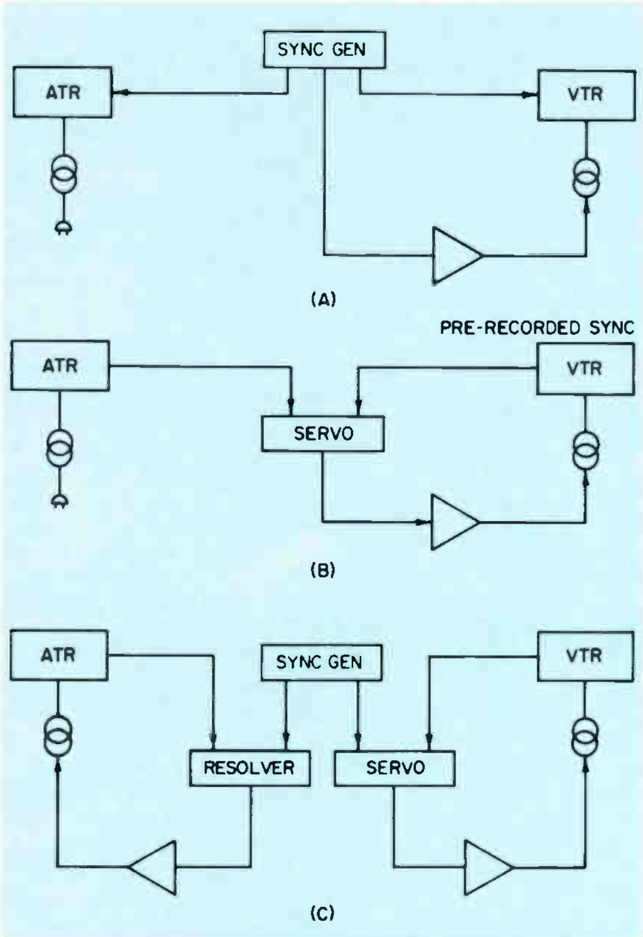



*Figure 10. The same designations for (A), (B), (C) as in Figures 8 and 9.*

in intermediate stages. It should be noted in this example that we have met the asymptotes for *slave* operation and they coincide with both the assumptions of primary *Rule 1* as well as *Rules 5, 6, and 7* for *slave* operation.

Other ways of obtaining synchronous operation are shown in the sketches of FIGURE 8 through 10. While they are self explanatory and the reader should find that they conform to the seven guidelines analzyed above, an additional word should be said about the final sketches of FIGURES 11 through 13.

FIGURE 11 shows an accepted means for playing back pre-recorded music to an actor who is being recorded on film as he performs to the music. (The machine recording the composite audio is not shown, only the one being used for playback). The audio technician trained solely in television may wonder why it is necessary for the reproducer to follow the camera speed, since in television we commonly let the playback machine take its own speed. In television we have, traditionally, used single-system recording whereby there is no need to keep the audio playback tape in sync with anything; all syncing is done by the actor, and once the composite sound is laid on video tape during the program shooting, that's the end of its use.

In those instances of double-system recording for television recommended in this discussion, the method for handling playback sound is detailed in Part 1, and that is also basically what is being done here in the film operation shown in FIGURE 11. It is enough to remember that the music playback tape will ultimately be transferred to sprocketed magstripe, which in turn will be locked to the picture on film. Therefore, the ratio of sound to picture must be maintained at all times. If the camera changes speed during shooting the musical number, the playback

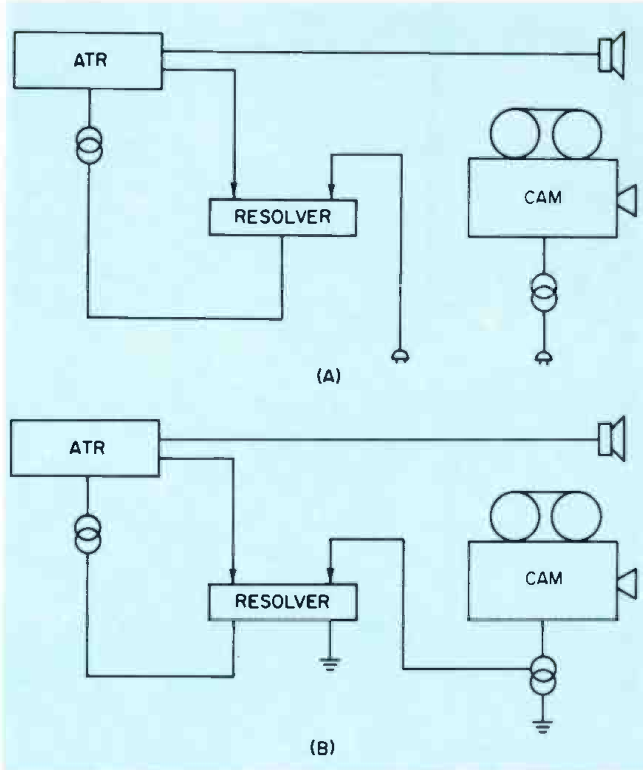


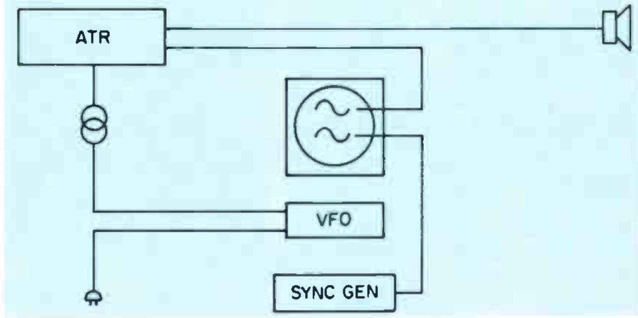

*Figure 11. An accepted means for playing back pre-recorded music. At (A) post-sync playback shooting (line); (B) post-sync playback shooting (battery).*

tape had better well change speed along with it, or we will have lost our reference of picture to sound, and will have failed to meet the requirements for synchronous operation as stated in *Rule 1*. Notice that the music playback tape has had a 60 Hz pulse recorded on it *prior* to shooting the picture. While FIGURE 11(A) shows *line* operation and FIGURE 11(B) shows *battery* operation, in each case the reproducer is referring to camera speed.

FIGURE 12 is a crude but most practical way of maintaining constant speed during repeated playbacks of a pre-recorded music tape. Again, a sync pulse (of any convenient frequency) is laid on the audio playback tape *prior* to shooting the picture. A dual-trace oscilloscope is fed both the output of the audio tape's sync track as well as the frequency source from which this track got its pulse. The playback tape machine is driven by a variable-frequency oscillator with course and fine tuning controls so that the operator can manually keep the two traces on the scope in line with one another. It is imperative that successive playbacks of such tapes are exactly the same in time, in order for the v.t.r. editor to use a part of one take with a part of another.

FIGURE 13(A) shows a general method whereby the

*Figure 12. A crude but practical way of maintaining constant speed during playback of a pre-recorded music tape. In short, manual resolving of a servo circuit.*



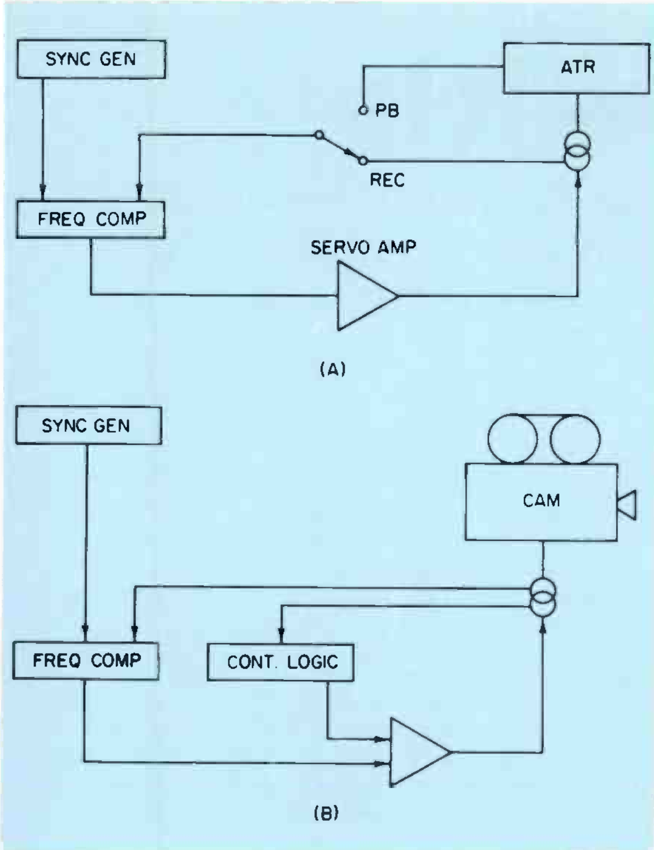



*Figure 13. A general method in which some manufacturers of audio tape machines solve the problems of double-system recording. At (A) the double-system tape recorder is shown, while at (B) the sync control for the film camera is detailed.*

more progressive manufacturers of audio tape machines have met the problem of double-system recording. A 60 Hz generator is self-contained in the recorder, and in the record mode the machine's speed is monitored by a ratio detector keeping a sharp eye on the tachometer wheel on the motor shaft. If the frequency generated by this rotating wheel doesn't match the frequency of the generator, the servo amp makes the correction. In the playback mode the ratio detector monitors the tape playback speed by noting the rate at which the recorded sync pulses pass a tachometer head. Again, if this doesn't match the 60 Hz of the generator, a correction is made as before. Thus, a strict 60 Hz operation can be maintained during both recording and playback.

Finally, the designers of certain motion-picture camera drives have embraced the idea just described, as generalized in Figure 13(B). They are incorporating a circuit similar to that of Figure 23(A) within the camera. The same frequency of 60 Hz has been settled on for reference, as this is compatible with the audio tape machines of Figure 13(A). A tremendous advantage here for film makers is that now several cameras can be shooting at once with only one recorder serving the bunch, and they'll all be in sync—an instance of film borrowing an idea from television.

The uses of sync pulses discussed herein are not claimed to be all-inclusive, but rather, typical and indicative. It is hoped by most technicians the author has met that some standardization of method is not far away. In the meantime, what we have done here is to examine the general nature of synchronous operation and to suggest some guidelines which hopefully will be applicable even in situations not specifically described. ■

R. H. CODDINGTON

# Live Rock: How It Is

*The author reviews the public-address equipment requirements of rock groups, including the sound-pressure level demands. He then offers some suggestions on a responsibility that sound men may not be shouldering.*

THE ROCK BAND CULT seems to be growing in volume—literally. The power of the amplified instrument is enfolding the *now* generation in a crescendo of escalating decibels that leaves long-established professional sound men groping in bewildered obsolescence. While it has been alleged that the trend in popular music now is toward more sane levels, corroborating evidence is hard to find.

The rock groups have made the scene long enough by now for the sound engineers in recording studios and major auditoriums to have adapted to their ultra-high-level acoustic requirements. However, with bands proliferating and their live appearances emigrating to the hinterlands—medium cities and the smaller college campuses—many uninitiated sound men are losing the battle of the decibels. A little sharing of experiences by those who have learned how it is could save these hapless sound men some red faces and jeopardized professional reputations. With that motivation, some observations are presented here.

Basic to understanding the phenomenon of rock music is the knowledge that the name of the game is *loud.* Deafeningly loud, but *also clean.* Unlike too many of the general public, who accept massive distortion as an inevitable characteristic of public-address systems, most rock musicians are sufficiently enlightened to know better.

The music must be loud because it is intended not to be *heard,* but to be *felt.* Failure to understand this fundamental tenet has frustrated many a seasoned adult seeking lower levels from the family stereo or the neighborhood 'teen band. Melody, rhythm, and lyric all are secondary to the sensual impact of an ambient sound pressure level (s.p.l.) that is virtually tactile. This—rightly or wrongly—is the appeal of the live rock band. The acceptance of this is the first step in the sound engineer's indoctrination to rock.

It is from this that further steps follow logically. For example, consider the singer, handicapped by the inherent relative weakness of the human voice. He somehow must rise above the din of the amplified instruments surrounding him. He has learned to do this by screaming into a microphone that is in intimate contact with his lips. At this distance, the human voice may generate s.p.l.'s in the 120-130 dB range, giving it a slight advantage over the incrementally lower ambient levels assaulting the microphone's diaphragm.

This has led to a new microphone technique that dismays seasoned sound and broadcast engineers. It also has spawned a new generation of "performer's" microphones that are designed to accommodate this technique without overloading. A supply of these (which may not be very suitable for more conventional sound jobs) is the first investment for the public-address engineer entering the rock band arena.

These microphones are designed with built-in blast- and pop-filtering, bass rolloff switches (in certain directional models) to offset the proximity effect, and acoustic circuitry to minimize diaphragm overload. Unfortunately, little has been done thus far to avoid electrical overload of the subsequent miscrophone amplifier; manufacturers generally are not yet incorporating switchable attenuators into their performer's microphones.

Overloaded amplifier input stages are a common problem when microphones are screamed into at point-blank range. The remedy is the insertion of fixed attenuators of at least 15 dB between microphones and amplifier inputs. Such pads, in a convenient in-line plug-in configuration, finally have appeared on the market. The rock-bound sound man will need a pocketful of these.

Having taken these steps to avoid microphone and pre-mixing overload, the sound man finally must look to his output power. The inexpensive 60-watt p.a. amplifier and a few paging-type horns simply will not suffice. He must be able to deliver a clean 110-120 dB s.p.l. or better throughout the audience area. This calls for top professional power amplifiers and speaker systems.

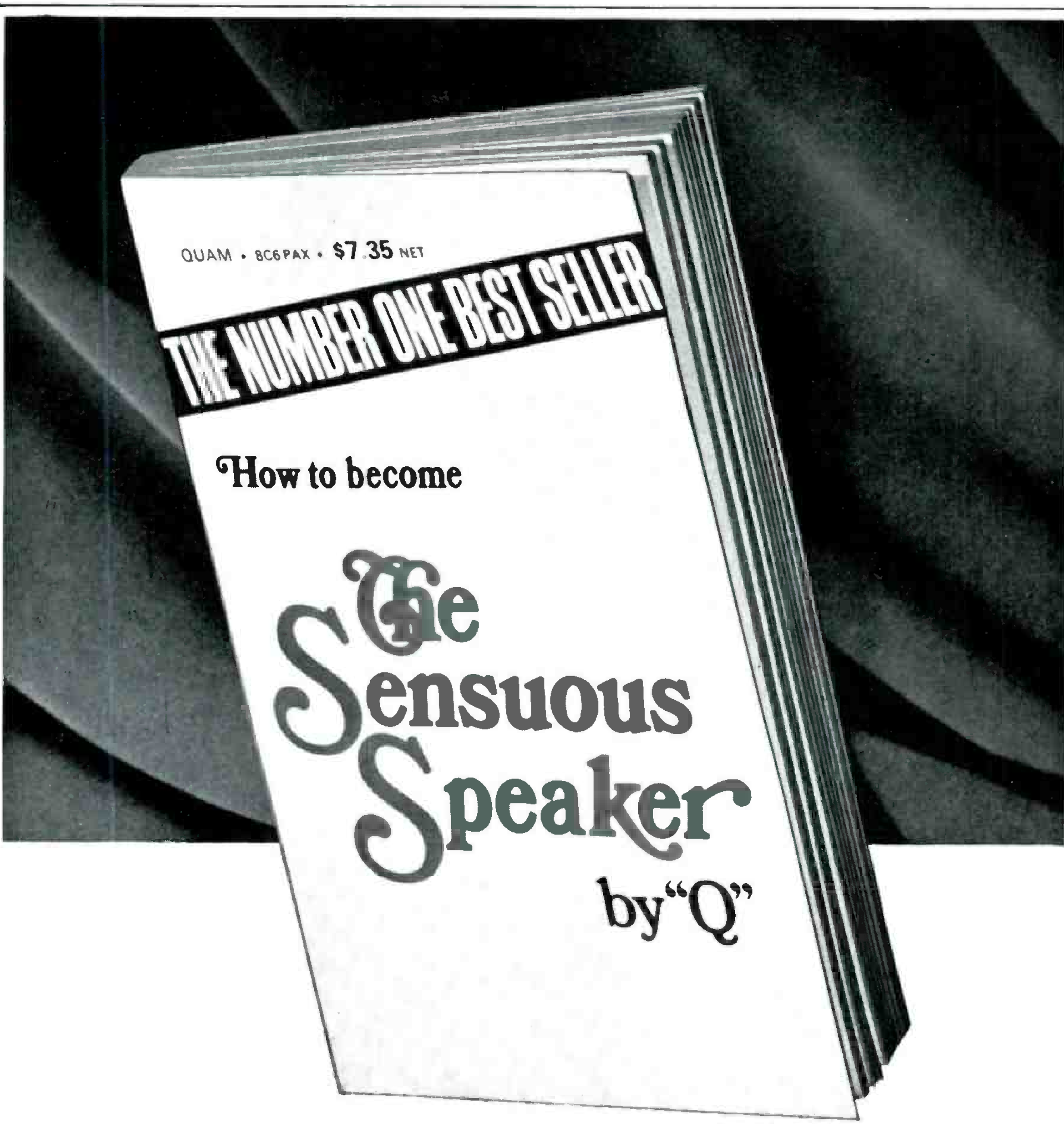
If only voice reinforcement is required, efficient cellular horns with a 300-hertz cutoff *may* suffice. In this case, a couple hundred or so watts may cover a small audience. When full-range enhancement of the instruments must be provided, however, much higher power is necessary to drive large bass enclosures.

These power requirements are illustrated on a small scale by a recent example in my experience. The location was a live gymnaisum/auditorium, about 150 by 150 by 30 feet. The group appearing was small, with only five instrument amplifiers on stage. These were to be amplified, along with group voices and occasional trumpet and saxophone passages, so full-range reinforcement was required. The mix involved twelve microphones.

In this example, the modest audience was covered with four semi-horn bass enclosures using fifteen-inch drivers, topped with multicellular horns and appropriate drivers. Of the horns, two were medium-throw and two were long-throw. Amplifiers totalling 500 watts of audio fed this speaker complex—and it proved to be none too much! (It has become axiomatic that *no* system ever has too much power; rock bands always use every dB available.)

***. . . many uninitiated sound men are losing the battle of the decibels.***

*R. H. Coddington is with Jarvis Co., Inc., a sound contractor in Richmond, Va. He is the author of Modern Radio Broadcasting, pubished by Tab Books.*






*Circle 33 on Reader Service Card*

*Is the sound engineer but a technician pursuing the narrow responsibility of equipment performance?*

One further facility provided is necessary to many rock band jobs: the monitor, or *sleeper* system. This is a speaker arrangement for the band members to hear themselves.

The four inches or so from lip to ear represents a loss path of several dB for the singer's voice, relative to the s.p.l. at the microphone. To hear himself above the ambient music level, that singer needs to overcome the lip-to-ear loss via a monitor system. In the example cited here, adequate monitor coverage was provided by 100 watts allocated to two column speakers aimed at either end of the bandstand.

Including the spill from the monitor system, the 600 watts employed in this case delivered a solid 110-113 dB s.p.l. uniformly through the auditorium—and it was working at capacity to do so. Anything less would have failed to meet the rock group's requirements.

At that, the requirements were modest. Some groups seek even higher levels. Had this job required as much as 116 dB s.p.l., evidently something over 1000 watts of audio would have been necessary.

From this experience, it can be seen that power requirements for the large auditorium, dance hall, or outdoor stadium become astronomical when a rock show comes to town. To meet these needs, knowledgeable members of the rock generation are forming their own high level p.a. companies. Their top-flight gear may include 2500 watts or more of audio in portable racks, highly flexible multi-channel mixing and equalizing facilities, and a variety of speakers to suit any job. Some of the more prosperous rock groups carry their own sound contractor and his equipment on road tours, but high level sound entrepreneuers also are establishing themsevles in the medium-sized cities, where they will be competitive with the long-established professionals.

A system capable of delivering clean high-level audio also sounds great at more conventional levels, of course. Operators of these systems must recover their substantial investments, which they will do by seeking—and winning—many of the non-rock functions that old-line sound men have served adequately and profitably in the past. This puts the long-established operator in the position of being forced to upgrade (or at least up-power) his equipment inventory, if he is to meet this new competition on an equal footing. This may seem a high price to pay just to stay in business, but there is one compensating factor: expensive equipment, operated with expertise, commands premium rates. Promoters of contemporary shows have learned (or are learning) that some shows refuse to appear without a guarantee of stipulated sound facilities, so they have little choice but to pay the freight. Even in the smaller towns, a good high-level sound system can command (depending on the facilities requested) from 350 to 1000 dollars for a one-night stand. These rates justify fine equipment, intelligently installed and operated.

Apart from the economics, there is a philosophical aspect ot this high-level sound business that will trouble some sound men. This arises from the acoustic levels involved; it has been demonstrated repeatedly that prolonged exposure to such levels invites permanent physiological ear damage. Thus, the current musical fad becomes a genuine hazard for the majority of young people. Will the p.a. enterpreneur close his eyes to his possible contribution toward a deafened generation for the sake of economic expedience? Or will he, knowing full well that the youngsters will seek to immerse themselves in destructive levels provided by *someone*, nevertheless forego today's profit for tomorrow's clean personal conscience? The choice is up to the individual.

Or is it? Is the sound engineer but a technician pursuing the narrow responsibility of equipment performance? Or is he a genuine professional, being obligated thereby to advise society in the areas of his expertise? Does his concern extend beyond his own direct participation, even to seeking restrictions on the excesses of others?

The general public is aware neither of the s.p.l.'s or the auditory hazards of live rock. Surely it's time for *someone* of professional authority to speak out on this aspect of noise pollution. Who is to do so, if the sound engineering fraternity stands collectively and individually mute?

Hearing specialists among the medical profession are sounding the alarm, but this is largely an academic voice that goes virtually unheeded by the general population. A concerted, official stand by the sound engineering profession could amplify this voice of warning and drive it home.

But can sound men do more than simply alert the public? Since they are at the controls, it might seem that they could actively resist the pressures for excessive sound levels. In practice, though, it soon becomes evident that this is futile; rock groups are noted for refusing to perform if they are denied the levels they desire. Too, the sound operator has no control over the band's own equipment, which is the fundamental offender.

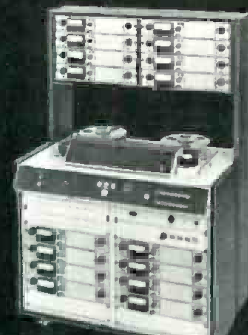
The alternative is for the sound fraternity to compaign for *legal* safeguards. Distasteful though it is to suggest even more laws in an era that already has far too many, the physiological hazards inherent in today's music seem to leave no workable alternative. Here is a clear mandate for professional sound organizations and societies to perform a vital service to tomorrow's society.

The first step is to *define*—with the help of physiological experts—safe and reasonable maximum s.p.l. standards for public performances. These should be widely disseminated, in layman's language, in all media. (The press release and the broadcast public-service announcement surely are as available to societies of sound professionals as they are to other non-profit organizations.)

This educational phase would lay the groundwork for the next step: using the prestige of the professional organization to press for legislation by local and/or state governments that would set legal s.p.l. limits for all public meeting places. This might be timely today, what with the growing public abhorrence of *other* forms of pollution, and the fact that non-rock generations still predominate in legislative bodies!

The objectives of this campaign would be for every meeting place open to the public to be permanently equipped with some form of sound-level metering and at least one individual charged with enforcing the legal s.p.l. limits at all functions. This certainly would incur the wrath of today's rock groups, who would be forced to conform or else be relegated to whatever private facilities they could command, but—if *all* public places were so governed—the current trend toward extreme s.p.l.'s ultimately should be reversed toward more reasonable values.

Admittedly, this is a pretty ambitious proposal. Such an objective may be beyond reach, but the effort could do much to awaken the public to the seriousness of the problem. Perhaps we at least might arrange for the placement of signs near the entrances to public places: *Caution: Entering these doors may be hazardous to your auditory health.* ■



# All together now...

Every beat, every note, tone, nuance faithfully reproduced. Every machine faithfully reproducing in sync. Ampex, leader of the 16-channel revolution, strikes again!

Now! The MM-1000 gives recording studios and teleproduction houses a new spectrum of creativity and quality. Now! The MM-1000 provides time and money-saving features offered by no other multichannel recorder.

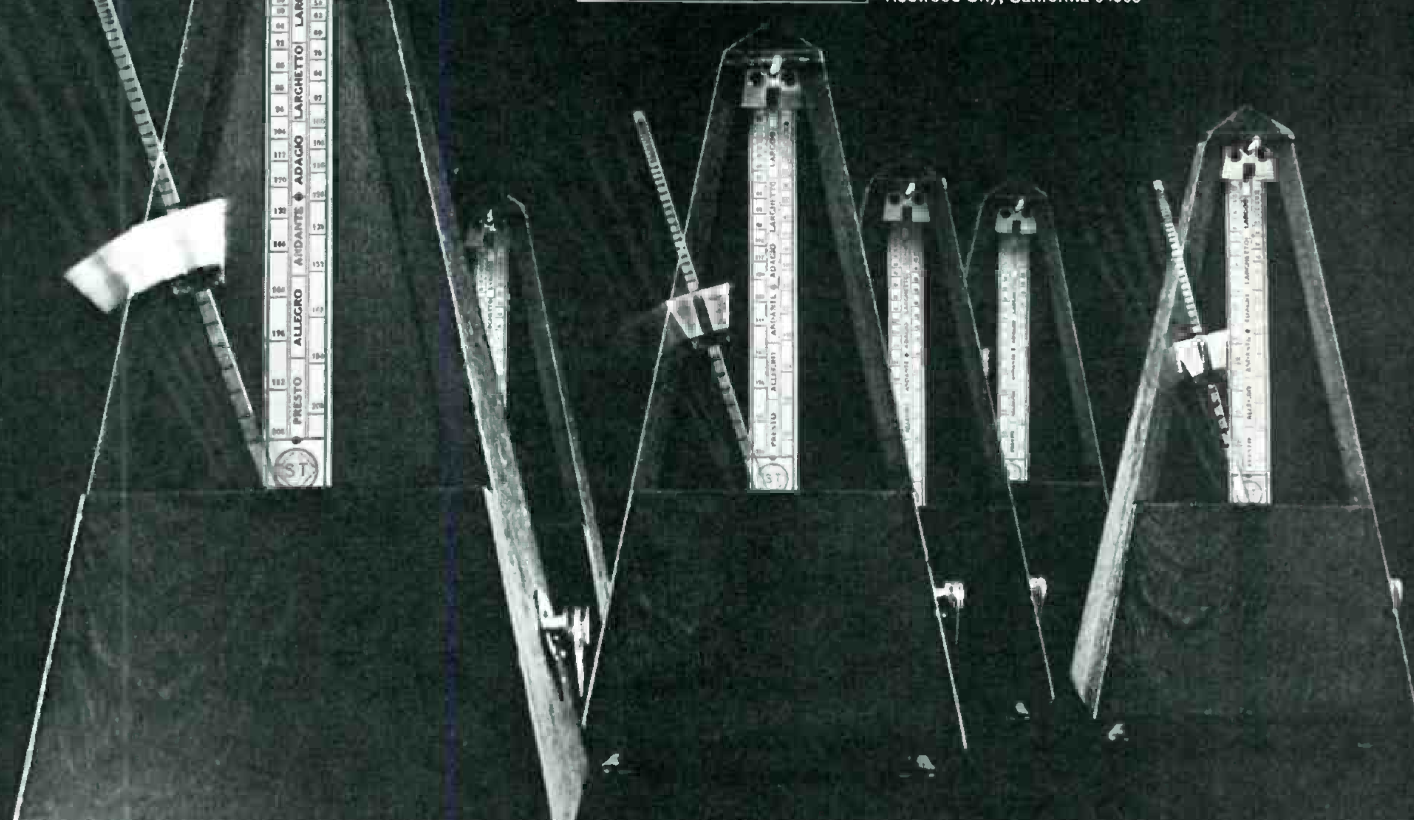
Recording studios can lay it down like never before with the MM-1000's exclusive Capstan Servo accessories. Through the reel timing accuracy. Precise pitch. Variable speeds. Every sound the same because record and playback are exactly matched: whether standard 7½, 15, 30 ips, or varied. *Plus,* 16 fully calibrated channels and the rugged reliability of the recording world's most popular multichannel recorder.

Teleproduction houses can attain multichannel sync like never before with the MM-1000's new Auditec System. Auditec, coupled with an automatic programmer provides direct synchronized linking of multichannel audio recorders with videotape recorders and station sync pulse. For the first time, multichannel audio can be recorded, programmed, and played back in automatic broadcast sync with other video and audio recorders. Producers and clients can change, correct, and approve synchronized sound and picture in one quick, convenient session.

Get right on with the MM-1000's. Ask your Ampex Representative about a Lease/Purchase Option, or write:

**AMPEX**

Ampex Corporation, Professional Audio Products Division
M.S. 7-13, 401 Broadway
Redwood City, California 94063



Attending the AES Convention October 5-8? Relax in the Ampex Gallery with the latest in MM-1000 investment protection PLUS something new for everyone.

*Circle 34 on Reader Service Card*

# 41st AES Convention and Exhibition

## Audio Engineering Society Exhibit

| | *Booth Nos.* |
|---|---|
| Acoustic Research | Rm 723 |
| AKG Microphones, North American Philips Corporation | 3/4 |
| Allison Research Inc. | 87 |
| Ampex Corporation | 92-98 |
| Audio Designs and Manufacturing Inc. | 78/79/80 |
| Automated Processes, Inc. | 58/59 |
| BASF Systems, Inc. | 13/14 |
| The R. T. Bozak Manufacturing Company | 19 |
| Burwen Laboratories | Rm 726 |
| CBS Laboratories | Rm 722 |
| Capps & Company, Inc. | 20 |
| CCA Electronics Corporation | 67/68 |
| Crown International | 49 |
| DBX, Inc. | Rm 735 |
| Dolby Laboratories, Inc. | 27/28 |
| Dukane Corporation | 42/43 |
| Electro-Sound, Inc. | 36/37/38 |
| Electro-Voice, Inc. | Parlor B & C |
| Elpa Marketing Industries, Inc. | 2 |
| Eventide Clock Works | 91 |
| Fairchild Sound Equipment Corporation | 88 |
| Daniel N. Flickinger & Associates, Inc. | 65/66 |
| Gately Electronics | 60/61 |
| Gotham Audio Corporation | Parlor A |
| Harvey Radio Company, Inc. | 26 |
| HAECO | 52 |
| Hitachi Maxell Ltd. | 5 |
| Inter. Telecom. Inc. | 76 |
| JVC America, Inc. | Rm 729 |
| Koss Electronics Inc. | 50 |
| Lipps, Inc. | 18 |
| Martin Audio Corporation | 57 |
| MCA Technology, Inc. | 89/90 |
| Multitrack | 85 |
| Nagra Mag. Recorders | 45/46 |
| Rupert Neve Incorporated | 55 |
| Olive Electro Dynamics Inc. | 71/72/73 |
| Ortofon | 62 |
| Otari of America Ltd. | 83 |
| Pentagon Industries, Inc. | 77 |
| Philips Broadcast Equipment Corporation | 33/34/35 |
| Pratt Sales Corp. | 12 |
| Quad-Eight Sound Corporation | 63/64 |
| RCA Commercial Electronics Systems | 15/16 |
| Recortec, Inc. | 70 |
| Revox Corporation | 17 |
| Samuels Engineering | Rm 724 |
| Sansui Electronics Corp. | Rm 736 |
| Scientific Electronic Systems | 1 |
| Scully Recording Instruments | 31/32 |
| Sennheiser Electronics | 21 |
| Shure Brothers, Inc. | 30 |
| Spectra Sonics | 74/75 |
| Stanton Magnetics, Inc. | 22 |
| Superscope, Inc. | 53/54 |
| Systron-Donner Corporation | 29 |
| TEAC Corporation | 56 |
| 3M Company | 39/40/41 46/47/48 |
| Tonus, Inc. | Rm 731 |
| United Research Laboratories Corp. | 51 |
| Vega Associates | Rms 718/719 |

## SCHEDULE OF EVENTS

### Hotel New Yorker

Monday, October 4—Welcoming Cocktail Party
6:00 to 7:30 P.M., Location will be posted

### REGISTRATION

Mezzanine

Tuesday, Oct. 5th—8:00 A.M. to 8:00 P.M.
Wednesday, Oct. 6th—8:30 A.M. to 8:00 P.M.
Thursday, Oct. 7th—9:00 A.M. to 5:00 P.M.
Friday, Oct. 8th—9:00 A.M. to 5:00 P.M.

### EXHIBIT HOURS

Mezzanine, 3rd and 7th Floors
**Tuesday and Wednesday,**
October 5 and 6—1:00 to 9:00 P.M.
**Thursday and Friday,**
October 7 and 8—11:00 A.M. to 5:00 P.M.

### TECHNICAL SESSIONS

**Terrace Room:** Sessions A, B, D, E, F, H, J, L, M
**New Orleans Room:** Sessions C, G, I, K

**Tuesday,** October 5
9:00 A.M.—Annual Business Meeting—Terrace Room
9:30 A.M.—A—Transducers
2:00 P.M.—B —Magnetic Recording and Reproduction
2:00 P.M.—C —Medical Electronics
7:30 P.M.—D—Digital Techniques in Audio

**Wednesday,** October 6
9:00 A.M.—E—Workshop on Studio Tape Recorders
2:00 P.M.—G—Acoustical Noise Control
7:30 P.M.—F —Special Evening: Care and Feeding of Tape Recorders

**Thursday,** October 7
9:30 A.M.—H—Disc Recording and Reproduction
9:30 A.M.—I —Audio Instrumentation and Measurements
2:00 P.M.—J —Design of Audio Transmission Systems
2:00 P.M.—K —Sound Reinforcement and Architectual Acoustics
7:00 P.M.—Social Hour—New Orleans Room
8:00 P.M.—Awards Banquet—Terrace Room

**Friday,** October 8
9:30 A.M.—L —Amplifiers and Signal Processing Devices
2:00 P.M.—M —Electronic Music
7:30 P.M.— —Electronic Music Concert—Terrace Room

### LADIES ACTIVITIES

A program of activities is planned. Ladies will meet at 9:00 A.M. each day for coffee before commencing the day's activities. Suite number will be posted.

Ladies Committee: Mrs. George W. Bartlett
Mrs. Donald J. Plunkett
Mrs. Dorothy H. Spronck

### SESSION A

Tuesday, October 5, 9:30 A.M.—Terrace Room

### TRANSDUCERS

Chairman: JAMES F. NOVAK
*Jensen Manufacturing Division, The Muter Company, Chicago, Illinois*

**Modulated Air Flow Direct Radiator Loudspeaker**—Harry F. Olson, RCA Laboratories, Princeton, New Jersey

**"Tunnel-Reflex": A Miniaturized Speaker System Using Folded Column Resonator Enclosures**—John J. Virva, Admiral Corporation, Chicago, Illinois

**Predicted vs. Measured Behavior of a Direct Radiator in a Vented Enclosure**—James V. White, Stevens Institute of Technology, Hoboken, N. J.
**Simplified Loudspeaker Measurements at Low Frequencies—Richard H. Small,** School of Electrical Engineering, The University of Sydney, Sydney, Australia
**Error-Free Passive Crossover Networks**—Allan L. Kaminsky, University of Colorado, Colorado Springs, Colorado
**Maxi-Bass and Marvelous-Midi from Mini-Woof-Woof**—J. Robert Ashley, University of Colorado, Colorado Springs, Colorado
**Miniature Transducers Illustrating Several Methods of Excluding Water**—George J. Sebesta and Richard W. Carlisle, Dyna Magnetic Devices, Inc., Hicksville, New York

## SESSION B

Tuesday, October 5, 2:00 P.M.—Terrace Room

### MAGNETIC RECORDING AND REPRODUCTION

Chairman: MARVIN CAMRAS
*IIT Research Institute, Chicago, Illinois*

**Velocity Sensing—The Parameter for a Complete Tape Transport Motion Control**—James C. Strickland, MCI, Inc., Fort Lauderdale, Florida
**The Stick-Slip Phenomenon at Low Magnetic Tape Speeds**—Robert Owen and George Benn, IIT Research Institute, Chicago, Illinois
**Some Considerations About Improvement of the Tone Quality of Cassette Tape Recorders**—Masahiko Morizono and Kenkichi Umeda, Sony Corporation, Tokyo, Japan
**Chromium Dioxide Audio Cassette Tape**—L. K. Jordan and J. E. Dickens, E. I. duPont de Nemours and Co., Wilmington, Delaware
**A Study to Establish Optimum Level on Cassette Copies**—Stewart L. Smith and James B. Wood, GRT Corporation and Audio/Tek, Inc., Sunnyvale, California and San Jose, California
**Dynamic Noise Limiter**—Edward R. Hanson, North American Philips Corporation, New York, New York
**A Real Time Spectrum Display for Master Tape Evaluation**—Tom Montgomery, Olive Electro Dynamics, Inc., Montreal, Quebec, Canada
**Design of a Noise Eliminator System**—Richard S. Burwen, Burwen Laboratories, Lexington, Massachusetts

## SESSION C

Tuesday, October 5, 2:00 P.M.—New Orleans Room

### MEDICAL ELECTRONICS

Chairman: PHILIP KANTROWITZ
*Consultant on Bioengineering, New York, New York*

**Innocent Murmurs in Children: Self Teaching Program for Physicians**—Roland Schmidt, West Virginia University School of Medicine, Morgantown, West Virginia
**Acoustic Evaluation of Prosthetic Cardiac Valve Performance**—Benedict Kingsley, Edward Waxler and Peter Green, Hahnemann Medical College & Hospital, Division of Cardiology, Philadelphia, Pa.
**Ultrasonic Diagnosis in Medicine**—George H. Myers, Riverside Research Institute, New York, New York
**Seeing With Sound**—Byron B. Brenden, Acoustical Minerals Exploration, Inc., Richland, Washington
**Computer Modelling for Predicting Optimum Anesthesia**—Robert Crane, Crane Bio-Medical Instruments Inc., Elmont, New York
**The Education and Role of the Audio Engineer in Bioelectronic/Medical Engineering**—Dean A. DeMarre, Technical Education Research Center, Northeastern University, Boston, Massachusetts
**Survey of Pressure Measurement Interfaces in Sockets**—Carl Mason, Veteran's Administration, Prosthetic Center, Bioengineering Research Service, New York and Philip Kantrowitz, Consultant on Bioengineering, New York, New York

## SESSION D

Tuesday, October 5, 7:30 P.M.—Terrace Room

### DIGITAL TECHNIQUES IN AUDIO

Chairman: RONALD W. SCHAFER
*Bell Laboratories, Inc., Murray Hill, New Jersey*

**A/D and D/A Convertors: Their Effect on Digital Audio Fidelity**—T. G. Stockham, Jr., Department of Computer Science, University of Utah, Salt Lake City, Utah
**Applying Digital Technology to Audio: Delay, Transmission, Storage, and Other Forms of Processing**—Barry Blesser, Massachusetts Institute of Technology, Cambridge, Massachusetts
**Principles of Digital Signal Processing**—R. W. Schafer, Bell Laboratories, Inc., Murray Hill, New Jersey
**Restoration of Old Acoustic Recordings by Means of Digital Signal Processing**—T. G. Stockham, Jr., Department of Computer Science, University of Utah, Salt Lake City, Utah
**Synthesis of Unlimited Vocabulary Speech Using a Computer Controlled Channel Vocoder**—W. H. Jamison, H. Blatter, and F. I. Zonis, RCA Laboratories, David Sarnoff Research Center, Princeton, N. J.

## SESSION E

Wednesday, October 6, 1971, 9:00 A.M. to 5:00 P.M.
Terrace Room

### WORKSHOP ON STUDIO TAPE RECORDERS

A day-long "hands on" approach on how to optimize mechanical and electrical functions of professional tape recorders. There will be a prepaid special registration fee of $5 for this workshop, and attendance will be limited to 100. A portfolio of supporting literature is included.

**Morning Session: 9:00 A.M. to 12:00 Noon**

—An overview of approaches to achieving a stable and constant speed tape path, with examples of how to determine and interpret electromechanical performance.
*Arthur Gruber, AEG Assocs., East Rockaway, N.Y., chairman*

—How to make meaningful field measurements and to interpret their significance, including practical alignment and adjustment of various electronic and electromechanical parameters of tape recorders.
*J. G. McKnight, Ampex Corporation, Sunnyvale, California*

—How motors get the tape "from here to there" effectively.
*Carl Bernsten, Ashland Electric Products Inc., N.Y.*

—Techniques for field setup and adjustment of heads to achieve optimum electrical and mechanical performance.
*Bill Morin, Nortronics Co., Inc., Minneapolis, Minnesota*

—The recording medium—how the physical and magnetic properties of tape interface with the recorder.
*Delos Eilers, Magnetic Products Div., 3M Co., St. Paul, Minn.*

**Afternoon Session: 1:30 to 5:00 P.M.**

Technical representatives of six manufacturers of studio tape recorders will explain and demonstrate setup and maintenance procedures of their particular machines. Attendees will have the opportunity to work in small groups with each machine, receiving an intensive briefing on specifics of each manufacturer's design and operational philosophy. Participating companies and their representatives are:

| | |
|---|---|
| *Ampex Corporation* | *Ted Johnson* |
| *MCI* | *C. Harned* |
| *Mincom Division, 3M Company* | *Bill Wilson* |
| *N. V. Philips' Gloeilampenfabrieken* | *Henry Van der Wal* |
| *Dictaphone, Scully Division* | *Bob Berliner* |
| *Studer-Franz AG* | *Stephen Temmer* |

## SESSION F

Wednesday, October 6, 1971, 7:30 P.M.—Terrace Room

### A SPECIAL EVENING: THE CARE AND FEEDING OF TAPE RECORDERS

*Neil Muncy, Suburban Sound Inc., Bethesda, Maryland, assisted by Bob Berliner, C. Harned, Ted Johnson, Stephen Temmer, Henry Van der Wal, Bill Wilson.*

### SESSION G

Wednesday, October 6, 2:00 P.M.—New Orleans Room

**ACOUSTICAL NOISE CONTROL**

Chairman: WILLIAM SIEKMAN
*Riverbank Acoustical Laboratories, Geneva, Illinois*

**Room Acoustics**—Ranger Farrell, Ranger Farrell and Associates, Irvington-on-Hudson, New York

**Noise Control for Studios**—Paul B. Ostergaard, Ostergaard Associates, West Caldwell, New Jersey

**Sound Insulating Windows**—Robert A. Hansen, Robert A. Hansen Associates, Inc., New York, New York

**Acoustical Floating Floor Systems**—Norman J. Mason, Mason Industries, Inc., Hollis, New York

**Building Design for High Noise Level Areas in Recording Studios**—Kenward S. Oliphant, Acoustical Consultants, Inc., San Francisco, California

**Acoustical Requirements Burolandschaften (Large-Area Offices)—The Necessity of Sound-Masking Systems**—Ernst-Joachim Voelker, Acoustical Consultant, Stierstadt/Taunus, Germany

### SESSION H

Thursday, October 7, 9:30 A.M.—Terrace Room

**DISC RECORDING AND REPRODUCTION**

Chairman: LAWRENCE SHAPER
*Dyna-Empire, Inc., Garden City, Long Island, New York*

**A Practical High-Frequency Trackability Test for Phono Pickups**—C. Roger Anderson and Paul W. Jenrick, Shure Brothers Incorporated, Evanston, Illinois

**Playback Losses and the Design of Wideband Phonograph Pickups**—James V. White, Stevens Institute of Technology, Hoboken, New Jersey

**A Multielement Stereo Cartridge for Derived Center and Room Ambient Surround Playback**—John J. Virva, Admiral Corporation, Chicago, Illinois

**A New Dynamic Feedback Stereo Cutter-Head with Associated Solid State Driving System**—Howard S. Holzer, Holzer Audio Engineering Corp., Van Nuys, California

**Record Changer Design Considerations**—Robert J. Hammond, V-M Corporation, Benton Harbor, Michigan

**Signal Synchronous Variable Pitch Computer**—Ronald Marcucci and Santo Galatioto, Capps & Co., Valley Stream, New York

**A Compatible Stereo-Quadraphonic (SQ) Record System**—Benjamin B. Bauer, Daniel W. Gravereaux, and Arthur J. Gust, CBS Laboratories, Stamford, Connecticut

### SESSION I

Thursday, October 7, 9:30 A.M.—New Orleans Room

**AUDIO INSTRUMENTS AND MEASUREMENTS**

Chairman: EMIL L. TORICK
*CBS Laboratories, High Ridge Road, Stamford, Conn.*

**Instrumentation and Methods for Violin Testing**—Carleen M. Hutchins, Catgut Acoustical Society, Inc., Montclair, New Jersey

**A Digital O-H METER**—Richard J. Fabbri and Edward J. Foster, CBS Laboratories, Stamford, Connecticut

**Weighted Peak Flutter Measurement: A Summary of the New IEEE Standard**—J. G. McKnight, Research Department, Ampex Corporation, Sunnyvale, Cal.

**Transmission Link Tolerances for Stereo/Mono Compatibility**—F. K. Harvey, Bell Telephone Laboratories, Inc., Murray Hill, New Jersey

**Automatic Detection of Impulse Noise**—George R. Kinzie, Jr. and Daniel W. Gravereaux, CBS Laboratories, Stamford, Connecticut

**Instantaneous Power Spectra—Fourier Analysis in Real Time**—Allan L. Kaminsky, University of Colorado, Colorado Springs, Colorado

**Tonebursts, Transients, and Troubles**—J. Robert Ashley and Thomas A. Saponas, University of Colorado, College of Engineering, Colorado Springs, Colorado

### SESSION J

Thursday, October 7, 2:00 P.M.—Terrace Room

**DESIGN OF AUDIO TRANSMISSION SYSTEMS**

Chairman: JOHN WORAM
*Vanguard Records, New York, New York*

**Comparative Stereophonic Listening Tests**—Carl Ceoen, Radiodiffusion-Television, Humbeek, Belgium

**A Simplified Console Monitor Mix System Using State of the Art Technology**—Tom Montgomery, Olive Electro Dynamics, Inc., Montreal, Quebec, Canada

**A Quadrasonic Mixdown Console with Visual Indication of Audio Distribution**—Don Richter, Automated Processes, Inc., Farmingdale, New York

**Multichannel Matrix Encoding**—Duane H. Cooper, University of Illinois, Urbana, Illinois and Takeo Shiga, Nippon Columbia Co., Ltd., Kawasaki, Japan

**Analysing Phase-Amplitude Matrices**—Peter Scheiber, Audiodata Company, New York, New York

**Noise Reduction for FM Broadcasting**—David Robinson, Dolby Laboratories, Inc., London, England

**Proposed Universal Encoding Standards for Compatible Four-Channel Matrixing**—R. Itoh, Sansui Electric Co., Ltd., Tokyo, Japan

### SESSION K

Thursday, October 7, 2:00 P.M.—New Orleans Room

**SOUND REINFORCEMENT AND ARCHITECTURAL ACOUSTICS**

Chairman: PETER W. TAPPAN
*Bolt Beranek and Newman Inc., Downers Grove, Illinois*

**Microphone Techniques for Improved Pick-up of Stage Performances**—Allan P. Smith, U. S. Naval Training Device Center, Orlando, Florida

**Microphone Considerations in Feedback-prone Environments**—Robert B. Schulein, Shure Brothers Incorporated, Evanston, Illinois
**Discrete Field Measurement of Howlback Probability in Microphones**—Daniel Queen, AmpliVox—Perma-Power Division, Chamberlain Mfg. Co., Chicago, Illinois
**Eleven Day Sound System for 325,000 People**—Robert F. Ancha, Ancha Electronics, Inc., Chicago, Illinois
**Acoustics and Sound System Design for Sao Paulo Exhibition Hall**—Wilfred A. Malmlund and Allan J. Rosenheck, Bolt Beranek and Newman Inc., Canoga Park, California and Nelson de Sampaio Bastos, Gradiente Electronics S/A, Sao Paulo, Brazil
**Acoustics and Sound Systems at the Walnut Street Theatre**—David L. Klepper and Larry S. King, Klepper Marshall King Associates, New York, New York
**Improved Electronic Background Noise Generation and Distribution Systems**—Thomas R. Horrall and David H. Kaye, Bolt Beranek and Newman Inc., Cambridge, Massachusetts

## SESSION L

Friday, October 8, 9:30 A.M.—Terrace Room

## AMPLIFIERS AND SIGNAL PROCESSING DEVICES

Chairman: SAUL WALKER
*Automated Processes, Inc., Farmingdale, New York*

**The Monolithic Balanced Modulation as a Versatile Audio Switching Element**—Walter G. Jung, A.I.I. Corporation, Baltimore, Maryland
**A Quantitative Comparison of Devices for Electronically Programmed Attenuation of Audio Signals**—Robert A. Moog, Moog Musonics, Inc., Williamsville, New York
**Automation as Applied to the Mixdown Process**—Wayne Jones, Olive Electro Dynamics, Inc., Montreal, Quebec, Canada
**An Ultraminiature Console Compression System with Maximum User Flexibility**—Barry Blesser, Massachusetts Institute of Technology, Cambridge, Mass.
**A Fail-Safe Audio Power Amplifier**—Martin Gittleman, Automated Processes, Inc., Farmingdale, New York
**Application of Sum and Difference Signal Processing in Stereophonic Broadcasting**—Eric Small, WOR-FM, New York, New York
**A Stereo Program Phase Checker**—James B. Wood, Audio/Tek, Inc., San Jose, California

## SESSION M

Friday, October 8, 2:00 P.M.—Terrace Room

## ELECTRONIC MUSIC

Chairman: DAVID FRIEND
*ARP Instruments, Division of Tonus, Inc., Newton Highlands, Massachusetts*

**A Composer's View of Mitsyn**—Robert P. Ceely, New England Conservatory of Music, Brookline, Mass.
**Digital Composition and Control of an Electronic Music Synthesizer**—Paul Conly and Allen Razdow, Harvard University, Cambridge, Massachusetts
**Recent Development in the Design of Voltage Controlled Oscillators for Electronic Music**—Robert A. Moog, Moog Musonics, Inc., Williamsville, N. Y.
**Preset Programming of Electronic Music Synthesizers**—David Friend and Jeremy Hill, ARP Instruments, Division of Tonus, Inc., Newton Highlands, Mass.
**Tempered Scale Generation From a Single Frequency Source**—Robert B. Cotton, Jr., Hammond Organ Company, Chicago, Illinois
**Large Scale Integration in Organ Design**—Ray B. Schrecongost, Hammond Organ Company, Chicago, Illinois
**Digital System for Realistic Organ Tone Generator**—Ralph Deutsch, Research and Technology Division, Autonetics; North American Rockwell Corp., Anaheim, California

## ELECTRONIC MUSIC CONCERT

Friday, October 8, 7:30 P.M.—Terrace Room

**Martin Dickstein**

# SOUND WITH IMAGES

## From Space to Earth

Although the first U.S. manned flight took place just ten years ago with a 15-minute sub-orbital flight, the manned-flight space program itself had actually been started in late 1958, and that came after years of technology associated with unmanned space flight. From all these years of new theories, experiments, discoveries, inventions and developments, many new technological advances have not only helped to further the space programs, but have also found their way into everyday living.

Some of the newest developments have had to do with ecology, aeronautics, medicine or just general fields of research with diverse applications. Others have dealt with optics, television, and other visual fields. Some may, but many may not, have direct or indirect bearing on the audio/visual field, but we thought a brief recap of some of the applications might prove interesting.

The advances necesitated in television in order to provide the great live color coverage in the recent Apollo moonwalks were obvious to all those who watched. Burnout-proof cameras, light-sensitive tubes capable of operating in virtually total darkness and under the worst climatic conditions—these and more are (or soon will be) available for surveillance or hazardous observations. Expensive, but available.

Weather forecasting, long-distance communications, and even earthquake foretelling have all benefitted from the advanced technology coming out of NASA studies. The last one mentioned makes use of the reflector left on the moon (by the astronauts) to bounce back radar signals originating on earth. The accurate measurement of the travel times shows up any wobble in the earth's rotation about the axis and can thus indicate possible internal turmoil leading to earthquake activity.

Study in noise ecology has received top priority, and methods for reducing jet noise during takeoff and landing are now under consideration. The aim is to reduce the level of the jet sound by at least 10 dB for the next 10 years until the roar comes down to the point where it wil be about the level of the average community background. At present, there is already under study a testing program for a newly completed 22,000 pound thrust engine whose noise level is expected to be 15 to 20 dB below comparable engines presently in use. Another ecological improvement in this new engine is the lack of any visible exhaust smoke.

Smog research is now under way making use of the instrumentation and techniques originally developed to study planetary atmospheres and origins of life in space. The technique will trace the photochemcial production of pollution and dispersion in the air. This may not be in the audio/visual field, but the eyes will certainly benefit.

Up until the Apollo 14 flight, the substance used to clear the visors of the astronauts had a time limit of about 30 minutes under the conditions where the helmet temperature fell below the dew point of their breath. In the Apollo 15 mission, a new substance was used which can make virtually any surface fog-free indefinitely. Two coats of this inexpensive material are applied to the surface and then buffed with a lintless cloth. The substance can be applied to a windshield, water-diving visor, lenses, mirrors, projection window glass and the protective glass or plastic windows used in closed-circuit all-weather protective housings.

A very simple method of using a computer, X-ray movies, sonar, and a 3-D movie readout is now under development to permit heart specialists to isolate two beats of a heart for repeated study as necessary for any period of time. Any part of the contraction or expansion of the heart walls can be stopped and held on the viewing screen to observe dead or scar tissue, for instance. This method will provide a more exact means to study a heart from all sides in completing isolation from all other parts of the body surrounding the heart. This is in contrast to the usual X-ray methods which provide only a two-dimensional view and are sometimes obstructed by other internal organs which show up on the X-ray plate. Since the final information is also available in information bits from the computer which has been programmed with "normal" heart information, the signals from the heart under study can be transmitted over normal computer transmission lines to another location for study and observation.

One development directly involved with the eye came as a result of an anticipated problem which never really materialized. Toward the beginning of manned flight into space, it was expected that severe *G* forces would prevent astronauts from being able to move sufficiently to control their ship. A method was devised whereby the men could maintain control of the craft by using their eyes. As it turned out, the astronauts never did have this problem of physical movement, but the original concept has now been applied to another use.

The sight switch, as it is called, is mounted on conventional eyeglass frames which are worn by quadraplegics or paralyzed individuals to control the movement of a battery-powered motorized wheel chair without the use of hands or feet. A low intensity beam of light is directed toward each of the patient's eyes. A sensitive photo-detector mounted ad-

*Figure 1. This motorized wheel chair is controlled solely by movement of the occupant's eyes. It is based on technology developed by space flight. Photo is courtesy of NASA.*

jacent to the light source, senses the amount of light normally reflected from the white part of the eye. When the eye is moved so that the darker iris cuts the beam, the decrease in reflected light is read by the detector and a signal is fed to the chair controls. The left eye directs stop/forward/reverse motion while the right one controls steering. Activation signals are sequential for start/stop/reverse motion. It was found that the eyelid and the eyeball have similar reflective properties so that blinking does not affect the operation of the switch control.

This same type of eye control can now be adapted and applied to operate mechanisms to turn pages, turning light on and off, controlling any electrical devices, or even with small modifications, connected to industrial machines and typewriters.

Many government, industrial and consumer applications have been and will continue to be found for the scientific findings and resulting technological advances coming out of space flight research. Such things as a brain sensor and radio transmitter system, developed for space medical research, seem to be applicable to treatment of schizophrenic mental patients. Ionizing space radiations and their effects on normal cells are now being studied for possible application in the study of cancer growth. Pictures, radioed back from the moon and Mars, have been enhanced by use of a computer, and the method used is now being applied to the study of rapid analysis of human chromosome pictures. Research and development in communications satellites has reduced the cost of a single trans-oceanic telephone channel from $16,000 to $600.

What is coming next is anyone's guess. Better telephone service? Film projectors that won't tear sprockets or lose loops? Slide projectors that won't jam? We'll have to wait and see. ■

**ARNOLD SCHWARTZ**

# THE FEEDBACK LOOP

● Last month we discussed generation of second harmonic distortion by a function generator. We found that if we feed a sine wave, $\sin \omega t$, into a black box whose output is the square of the input signal, the output frequency will be double ($\cos 2\omega t$) that of the input signal. Harmonic generation is not the most objectionable component of distortion. Musical instruments generate fundamental tones and a series of harmonics, so that although the generation of harmonics by an audio device may alter the overtone structure of the music, it will not introduce discordant sounds usually associated with distortion.

Intermodulation distortion, however, does produce sounds that are unrelated to the harmonic series generated by the musical instruments themselves. The operation of the square function on a complex signal shows how intermodulation distortion is generated. In the simplest case we can assume two signals present at the input of our square function generator, $\sin \omega_1 t$ and $\sin \omega_2 t$. We are assuming that we have a linear system up to the input of the square generator so that each of these signals exists independently of the other. The input signal than is

$$(\sin \omega_1 t + \sin \omega_2 t)$$

The output of the square function generator will be the square of the input, Expanding this expression we have

$$\sin^2 \omega_1 t + 2(\sin \omega_1 t)(\sin \omega_2 t) + \sin^2 w_2 t$$

The first and last terms of this expression are the familiar ones derived last month and represent the generation of second harmonic of each of the fundamental inputs. The term $2(\sin w_1 t)(\sin \omega_2 t)$ is somewhat different. Using the appropriate trigonometric manipulation we find that

$$2(\sin \omega_1 t)(\sin \omega_2 t) = \cos(\omega_1 t - \omega_2 t) - \cos(\omega_1 t + \omega_2 t)$$

Combining all the terms

$$(\sin \omega_1 t + \sin \omega_2 t)^2 = \frac{\cos 2\omega_1 t + \cos 2\omega_2 t + 2}{2}$$

$$\cos(\omega_1 t - \omega_2 t) - \cos(\omega_1 t + \omega_2 t)$$

For those interested, trigonometric manipulations of this type can be carried out most easily by using the exponential form of the trigonometric functions

$$\sin x = \frac{e^{ix} - e^{-ix}}{2i}$$

$$\cos x = \frac{e^{ix} + e^{-ix}}{2}$$

The expressions $\cos(\omega_1 t - \omega_2 t)$ and $\cos(\omega_1 t + \omega_2 t)$ are the sum and difference signals, or intermodulation distortion. As stated above, harmonic distortion generated by a square function tends to blend with the overtones of the original musical signal. Intermodulation distortion is entirely different. Let $\frac{\omega_2}{2\pi}$ be 200 Hz and $\frac{\omega_1}{2\pi}$ be 321 Hz. The signal $\left(\frac{\omega_1}{2\pi} - \frac{\omega_2}{2\pi}\right)$ would represent a frequency of 119 Sz and the signal $\left(\frac{\omega_1}{2\pi} + \frac{\omega_1}{2\pi}\right)$ would represent a frequency of 521 Hz. Both sum and difference signals are not musically related to the original signal at the input of the square function generator. With a complex musical signal containing many fundamentals and related musical overtones, the intermodulation products can represent a substantial percentage of the original signal. These sum and difference signals, for the most part, should be considered as noise that has been combined with the original musical signal.

Non-linear systems can be divided into three catagories: those that generate only even order harmonics (second, fourth, sixth harmonics), those that generate odd order harmonics (third, fifth, seventh harmonics), and those that generate both even and odd order harmonics. Third harmonics (and the associated intermodulation distortion) is generated by a cube generator. The trigonometric expression for the cube generator is

$$\sin^3 \omega t = \tfrac{3}{4} \sin \omega t - \tfrac{1}{4} \sin 3\omega t$$

We can contrast three features of the cube generator with the square generator the ratio of distortion to input signal is lower, there is no d.-c. component, and a fundamental tone is generated. The reduction in the magnitude of distortion as the harmonic number increases is generally true of non linear systems. For this reason the total distortion of the system can be presented with reasonable accuracy by a square and cube generator, where both types of distortion are present. If only odd or even order distortion are present then the system can be represented by either the cube or the square generator. ■

# CLASSIFIED

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

Rates are 50¢ a word for commercial advertisements. Non-commercial and employment offered or wanted placements are accepted at 25¢ per word.

## FOR SALE

SOLID-STATE AUDIO PLUG-IN OCTAL (1" Dia. x 2" H) modules. Mic preamps, disc & tape preamp-equalizers, tape bias osc & record ampl power amps & power supplies. Send for free catalog and audio applications. **Opamp Labs., 172 So. Alta Vista Blvd., Los Angeles, California 90036.**

NEW YORK'S LEADING supplier of professional recording equipment and hi-fi stereo components. All major brands in stock. Call for quote—sales—service—leasing—trade-ins. **Martin Audio, 320 West 46th Street, New York, N.Y. 10036. Telephone: (212) 265-6470.**

AMERICA'S LARGEST SELECTION of new and used broadcast and recording equipment! Latest bulletins available. **The Maze Corporation, P.O. Box 6636, Birmingham, Ala. 35210.**

WHATEVER YOUR EQUIPMENT NEEDS—new or used—check us first. Trade your used equipment for new. Write for our complete listings. **Broadcast Equipment & Supply Co., Box 3141, Bristol, Tenn. 37620.**

HAECO announces complete repair service and overhaul for all Westrex cutterheads. Conversions of 3D-II and older models to higher performance standards and reliability. Helium cooling systems and hi-temp coils can protect your investment. Repair insurance program available. Rapid service. Lower cost. **HAECO, 14110 Aetna, Van Nuys, California 91401.**

RECONDITIONED TAPE REELS. Scotch precision or equivalent as new, 1 x 10½ inches—10/$35; 2 x 10½ inches—10/$70. Send for list: **Wide Response, 2926 Bentley Avenue, Los Angeles, California 90064.**

STUDIO MONITORS (8 units) sold in pairs only. Specially made for AES show display. $500.00 per pair, f.o.b. our plant. Call **Miss Dufine, Rectilinear Research Corp., (212) 585-9400. 107 Bruckner Blvd., Bronx, N.Y. 10454.**

SCULLY 16-track sync master control unit with 40-feet cable. Best offer. Call **Sigma Sound Studios, 212 N. 12th Street, Philadelphia, Pa. 19107. (215) 561-3660.**

TECHNICAL CONSULTATION—recording/broadcasting studios, and performing arts. Thirty audio lines—consoles, microphones, equalizers, monitoring systems. Call or write—quotations, specifications. Listings. **Frankford/Wayne Laboratories, 212 N. 12th, Philadelphia, Pa. 19107. (215) 561-1794.**

STUDIO CLEARANCE SALE #3. Ampex AG-300-8-C Solid State 8 track with sync. panels, console cabinet, using AG-350 electronics $5995.00. Ampex AG-350 full track recorder with two track stereo playback through McIntosh C-22 rack mount preamp. $1095.00. Ampex 300-2 two track stereo recorder with 350 electronics and new heads $1095.00. CROWN SS-722 with case and extras $695.00. Teletronix LA-2A leveling amplifiers $225.00 ea. Altec 1567A mixers with 3 transformers and meter $195.00 ea. Pultec MAVEC preamplifier equalizers $185.00 ea. A.K.G. Norelco C-60 System $135.00. Ampeg B-18 Bass amplifier $169.00. Ampeg Gemini II $169.00. 16 position solid state 8 track recording and remix console system consisting of Melcor and Altec components. $4395.00 For additional information write or call **Wiegand Audio Laboratories, R.D. 3., Middleburg, Pa. 17842. Phone (717) 837-1444**

OLD RADIO PROGRAMS on cassettes or reels. Thousands of your favorites live again. Low prices, high quality. Catalog 50c. **Remember Radio Inc., Box 2513, Norman, Oklahoma 73069.**

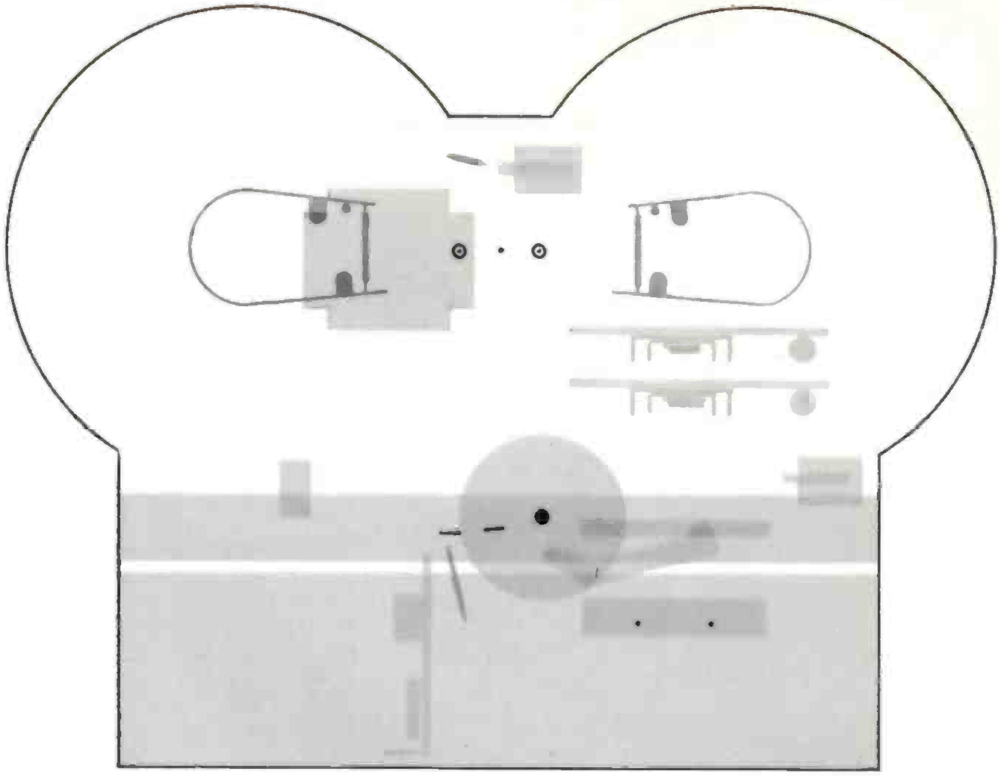
ONE STOP FOR ALL your professional audio requirements. Bottom line oriented. **F.T.C. Brewer Company, P.O. Box 8057, Pensacola, Florida 32505.**

## EMPLOYMENT

PROFESSIONAL RECORDING PERSONNEL SPECIALISTS. A service for employers and job seekers. Call today! **Smith's Personnel Service, 1457 Broadway, N.Y.C. 10036. Alayne Spertell 212 WI 7-3806.**

**Want to Live in Israel?** Rapidly expanding design, manufacturing, and sales company specializing in audio, high fidelity, commercial and military sound systems requires experienced, mature professionals in all of the above fields. For full details send complete resume or contact **L. Feldman, 97 Oxford Blvd., Great Neck, New York 11023. (516) 482-5629.**

DISTRICT SALES MANAGER WANTED. Leading AAA1 manufacturer of audio communication products offers outstanding opportunity to a salesman experienced in commercial sound and communication sales. Successful applicant will be responsible for building sales volume in multi-state area through effective selection, training and supervision of commercial sound distributors. Requires solid technical understanding of the field, willingness to travel extensively. Attractive compensation plan provides base salary and expenses, plus commission and bonus. Send your resume in full confidence to **Carl Dorwaldt, Rauland-Borg Corporation, 3535 West Addison Street, Chicago, Illinois 60618.**



# The new Revox A77 Mk III.
## It's still not perfect.

Nothing is.

But the new A77 Mark III is certainly the best recorder Revox has ever made. And that's saying something.

The Mark III is an improved version of our critically acclaimed A77. The recorder that The Stereophile magazine (1-71) described as, "Unquestionably the best tape recorder we have ever tested . . ."

And that judgement is as true now as it was then.

However, at Revox we've never been content to rest on our laurels. We thought we should make the best even better.

But in bringing out a new model, we didn't discard all of the time tested features and superior performance that distinguished the original A77.

Instead, we made only those changes which would meaningfully improve performance and reliability.

Not a radical transformation, but a program of rational development.

As a result, you have to examine the new A77 Mark III rather closely before you see any external differences at all.

On the other hand, from the moment you start to use the new Revox, you'll begin to appreciate the changes we've made inside.

For example, we've designed a new oscillator circuit for greater efficiency and lower distortion. Modified and strengthened the self-adjusting braking system. Devised a new hardening process to reduce capstan wear. Improved tape handling and spooling. And made a number of other changes. A total of eighteen . . . some major, some minor.

All in all, we haven't created a revolution.

We've just done what we set out to do . . . that is carry the art and science of tape recording a few steps closer to perfection.

And, in the process, we've given you eighteen more reasons why . . .

**REVOX**
delivers what all the rest only promise.

**Revox Corporation**
155 Michael Drive, Syosset, N. Y. 11791
3637 Cahuenga Blvd., West, Hollywood, Calif. 90068
In Canada: Tri-Tel Associates, Ltd., Toronto, Canada
Lamb House, Church Street, Chiswick, London W4 2PB

*Circle 11 on Reader Service Card*

# The new Koss Red Devil is one hell of a phone

**Model KRD711**
**only $29.95**

Use it for studio applications.

It is light weight, comfortable, practically blowout proof.

It presents no impedance matching problems.

It sounds better than any other phone you've heard in its price range.

For more information write KOSS
KOSS CORPORATION 4129 N. Pt. Washington Rd., Milwaukee, Wisc. 53212

*Circle 12 on Reader Service Card*