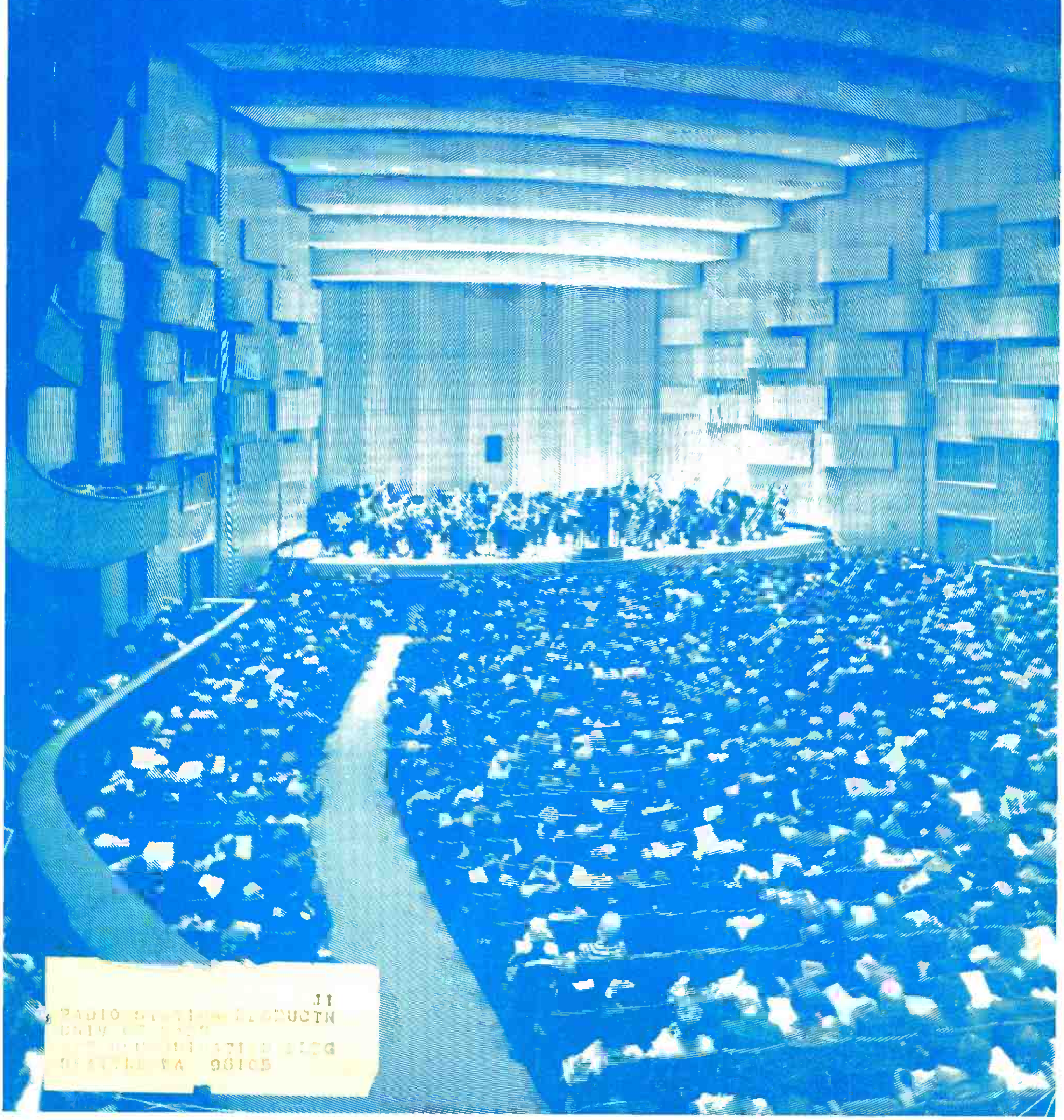
db

THE SOUND ENGINEERING MAGAZINE
MARCH 1971 75¢

Acoustics



The Dolby 360 Series

Nearly a thousand of these new units are already in use.

Each Series 360 unit is only 1¾ inches (44 mm) high. 16 channels therefore require only 28 inches of rack space.

Full compatibility with the A301

Models 360 and 361 are single-channel A-type (professional) noise reduction units which process signals identically to the two-channel A301. The new units are small in size and are designed for simplified installation and use of the Dolby System with 16-track recorders. The cost of the 360 series is somewhat less than that of the A301 for an equivalent number of channels.

Automatic record/play changeover in the 361

The Model 360 is a single-channel noise reduction processor unit. The Model 361 is identical to the 360 in size and appearance, but contains facilities for automatic record/play changeover controlled from the recorder. In the new series, the operating mode is set and clearly displayed by illuminated push-button switches.

Internal oscillator An internal "Dolby Tone" oscillator is provided for establishing correct operating levels. The characteristic modulation of the tone also identifies Dolby-processed tapes. All oscillators in a multi-track installation can be controlled by a single switch.

High stability The circuit is highly stable and does not require routine adjustment. A removable front panel allows input and output levels to be adjusted from the front of each unit. The panel also provides access to relays and the noise reduction module.

Single-module design The noise reduction circuitry is contained in a single module which can be purchased separately. Should failure ever occur, plug-in substitution will restore operation of the system in seconds with no adjustments necessary.

Prices, delivery information and complete specifications are available from

DOLBY LABORATORIES INC

333 Avenue of the Americas New York NY 10014
(212) 243-2525 cables: Dolbylabs New York

UK and International
346 Clapham Road London SW9
(01) 720-1111 telex: 919109 cables: Dolbylabs London

COMING NEXT MONTH

• A special issue devoted to microphones.

John Eargle has prepared a study of *How Capacitor Mics Produce Cardioid Pickups*.

Roger Anderson and Robert Schulein are co-authors of a paper discussing a new distant micing technique that is at once unique and simple. They tell how it is done and why it happens.

Our microphones were turned on at a symposium held between several recording engineers and a microphone manufacturer as represented by a senior development engineer. It's a stimulating and educational discussion you won't want to miss.

In addition, we will have a complete guide, map included, and the papers to be given at the *40th AES Convention and Exhibition* to be held at the L.A. Hilton in California. Dates are April 27th through the 30th.

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

• Illustrating the major topic of this issue, is this striking photo of Philharmonic Hall at Lincoln Center in New York City. The New York Philharmonic Society undertook the redesign of the original hall which had proved to be acoustically less than satisfactory.

Heinrich Keilholz, famed West German designer, was responsible for the new look and the new sound—estimated to cost $1.3 million. The photograph was taken by Sandor Acs and was supplied through the courtesy of the Lincoln Center for the Performing Arts, Inc.

db

THE SOUND ENGINEERING MAGAZINE

MARCH 1971 VOLUME 5, NUMBER 3

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 75c 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

The Editor:
In your December issue, the comments of R. N. Andrews in his article concerning integrated circuit operational amplifiers is not consistent with previously published material. Mr. Andrews indicates that descrete operational amplifiers have a noise of -129 dBm and an output of +19 dBm into 100 ohms. This compares with a noise of -127 dBm and an output of +22 dBm for buffered integrated circuit operational amplifier. Here the dynamic range of the IC circuit is greater which in this era of high output mics is what really counts.

Further, if one assumes that the noise measurements on the descrete operational amplifier were made using the circuit shown in *Figure 2*, then the information is still further distorted. This circuit presents a terminating rather than bridging impedence to the microphone, which while improving the measured noise figures would reduce microphone output. If a buffered IC microphone preamp were measured using a terminated arrangement rather than the preferred bridging input arrangement, it too, would have a noise figure of -129 dB giving the IC a full 3 dB improved dynamic range.

John H. Buffington
Chief Engineer
Gately Electronics
Havertown, Pa.

advertisers index

db
THE SOUND ENGINEERING MAGAZINE

SALES OFFICES

New York
Specifax Media Sales, Inc.
Manfred W. Meisels
603 East 82nd Street
New York, N. Y. 10028
212-249-7768

Boston
Specifax Media Sales, Inc.
William H. Fields III
49 Gray Street
Boston, Mass. 02116
617-426-5315

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

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

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

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

Pennsylvania
Specifax Media Sales, Inc.
Andrew N. McCreight
Box 199
Chester Springs, Pa. 19425
215-363-7355

An awesome problem:

You are the sound man for a rock group. And within a moderate budget you must provide high-level reinforcement of both instruments and vocals. But this is no ordinary rock band. In addition to the usual guitar-bass-drums, the group has a second drummer plus trumpet, trombone, and french horn.

There's more. You must also be able to cleanly reproduce english horn, clarinet, bass trumpet, saxophone, oboe, and flute. Because this rock group also plays the classics: Bach, Lully, Erb, Debussy.

A most unusual group, the Symphonic Metamorphosis. Eight top-rank musicians from the Detroit Symphony. With a creative approach to the generation gap: "fusion rock" that mixes classical music and hard-driving rock on the same program.

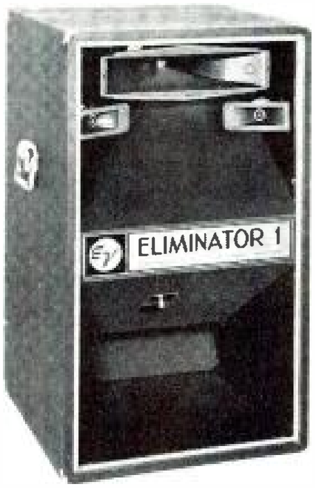
It makes a serious sound problem. From ear-splitting rock to a delicate flute solo. And it all must satisfy the critical ears of eight symphony musicians. Despite acoustic environments that range from beautiful symphony halls to atrocious meeting rooms. Or worse!

The Symphonic Metamorphosis records for London records.

The simple solution:

The Symphonic Metamorphosis bought Electro-Voice. Not on the basis of what we claim, but rather on what they heard. Using RE-11 and 635A microphones and just a pair of Eliminator I loudspeakers. They heard remarkable clarity, excellent separation of instruments, wide dispersion of sound, and the ability to reproduce the full dynamics of their unusual musical style. In short, the sound system fully complemented their serious approach to music.

Not all of your customers will make such uncompromising demands for sound. But all can benefit from the resources built into every Electro-Voice microphone and speaker. It would be a shame to offer them less.



Electro-Voice®

a **GULTON** subsidiary

ELECTRO-VOICE, INC., Dept. 1208D, 610 Cecil Street, Buchanan, Michigan 49107

Circle 19 on Reader Service Card

George Alexandrovich

THE AUDIO ENGINEERS HANDBOOK

• Like it or not, every bit of reproduced sound depends on performance of mechanical drive system—be it the capstan drive in a tape machine, the turntable drive system in reproduction from discs or vtr tape drive. A decade ago, when the record industry was booming, a lot was said, written, and discussed about different drive systems; belt drive *versus* idler or puck drive, direct drive from the motor, or clutch drive with damper. Well, most of this is now forgotten, except that equipment that was installed then has to be now serviced. Newcomers to the field, young engineers and technicians, should inherit some of the know how obtained through the blood and sweat of those who were designing this equipment and thereby be taught some of the basic facts about drives and their maintenance.

Let's start with classification of the drives: Simplest, is the *direct drive* where the motor shaft is ground to the correct diameter and acts as a capstan to pull the tape. Second, is the combination of the *motor driving* a larger diameter *flywheel* by directly engaging the resilient outer surface of the flywheel. Third, is the same motor engaging a larger wheel through a small *resilient idler wheel*. Last, is *belt drive*.

Direct-drive for tape recorders is one of the most reliable systems. The major advantage is that few parts can go bad. This type of drive is only suitable for tape machines because the motor has to be rigidly mounted on the deck, thus transmitting all of its vibration noise to the entire assembly.

Preventive maintenance of a direct drive consists of cleaning the capstan and pinch roller of tape oxide particles and dirt. Oxide is usually lubricated. If this lubricant (with the oxide particles) get deposited on the pinch roller in a larger quantity it may cause tape to slip. Uniform motion of the tape or rotational speed of the disc are vital to flutter-free reproduction of sound. In most professional tape machines the pinch roller disengages automatically when the machine is stopped or turned off. But less expensive machines may have their capstan roller engaged although power is off. Constant pressure on a roller deforms the rubber surface. Once the machine is restarted, the uneven surface of the roller results in uneven tape speed. Some rollers are made of good memory material (silicon rubber) while others may be made of neoprene or lower-grade materials. With roller made of poorer rubber it may be necessary to have it reground or replaced to again obtain constant tape speed. Sometimes, just running the machine for a while may help get rid of puck indentation. Try it, at least, before regrinding or changing.

In smaller machines such as cartridge or cassette units, the motor physically can not be placed next to the tape heads because it introduces hum. Another reason why direct drive is not used in small tape decks is that when pulling tape at 7 1/2, 3 3/4 or 1 7/8 in./sec., either the motor has to be of slow speed or its shaft has to be only several thousands of an inch in diameter—both conditions that are highly impractical. Therefore, the capstan is made to be the shaft of a flywheel that is driven by the motor directly at the rim or through the belt. Since direct rim drive appears only in a small number of designs, we will consider belt drive at this time.

The motor has small diameter pulley; the flywheel has a diameter several times larger. Speed constancy is accomplished through the use of synchronous motor or by means of governor if it is a d.-c. motor. Theoretically, a ratio of two wheel diameters will tell us the speed with which the flywheel will rotate. If motor speed is 1800 rpm and pulley diameter (or radius) ratio is 10/1, then the flywheel should rotate at 180 rpm.

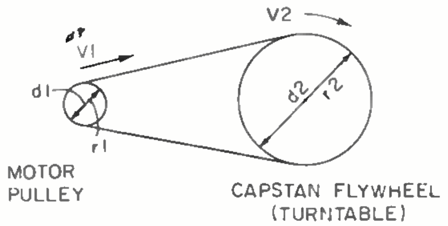
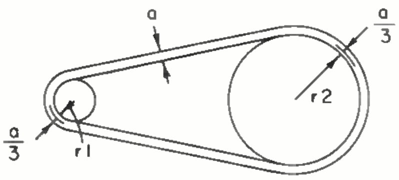


Figure 1. The ideal pulley diameter/speed relationship. V—rotational speed in rpm; d—diameter; r—radius. $\frac{d_2}{d_1} = \frac{V_1}{V_2} = \frac{r_2}{r_1}$.

Figure 2. A practical pulley/speed relationship. a= thickness of a flat belt. $\frac{r_2 + a/3}{r_1 + a/3} = \frac{V_1}{V_2}$.



To call it "an amplifier" would be like calling a Porsche "Basic transportation."

There is unusual satisfaction that comes from fulfilling a prosaic task in a far from prosaic manner.

Hence this amplifying system: the Sony TA-2000 professional preamplifier and the Sony TA-3200F power amplifier. Together, they perform all an amplifier's standard tasks in a satisfyingly impeccable manner; but their 67 levers, switches, meters, knobs and jacks allow you to perform some interesting functions that are anything but standard.

Dual-purpose meters.

The two VU meters on the preamplifier front panel, for example, are no more necessary than a tachometer on an automobile. But they do serve the dual purpose of simplifying record-level control when the TA-2000 is used as a dubbing center, and of allowing you to test your system's frequency response and channel separation (as well as those of your phono cartridge), and to adjust the azimuth of your tape heads.

A broadcast/recording monitor console in miniature.

The TA-2000 resembles professional sound consoles in more than its VU meters. In addition to the 20 jacks and seven input level controls provided on its rear panel for permanent connections to the rest of your hi-fi system, the TA-2000 boasts a professional patch board in miniature on its front.

Thus, you can feed the inputs from microphones, electric guitars, portable recorders or other signal sources into your system without moving the preamplifier or disturbing your normal system connections in the least. And a front-panel Line Out jack feeds signals for dubbing or other purposes into an external amp or tape recorder, with full control of tone and level from the front-panel controls and VU meters.

The tone correction and filtering facilities are also reminiscent of professional practice, allowing a total of 488 *precisely repeatable* response settings, including one in which all tone controls and filters are removed completely from the circuit.

The amplifier—no mere "black box"

A power amplifier can be considered simply as a "black box" with input and output connections, a power cord, and an on/off switch; and such an amplifier can perform as well (or poorly) as the next one. But in designing the TA-3200F Sony took pains to match the amplifier's facilities to the preamplifier's.

Thus to complement the TA-2000's two pairs of stereo outputs, the TA-3200F has two stereo pairs of inputs, selected by a switch on the front panel. Other front panel controls include independent input level controls for both channels, a speaker selector switch, and a power limiter (in case your present speaker should lack the power handling capacity of the next one you intend to buy).

Circuitry unusual, performance more so

The single-ended, push-pull output circuitry of the TA-3200F amplifier is supplied with both positive and negative voltages (not just positive and "ground") from dual balanced power supplies. This system allows the amplifier to be coupled directly to the speakers with no intervening coupling capacitors to cause phase shift or low-end roll-off (A switch on the rear panel does let you limit the bass response below 30Hz if you should want to, otherwise, it extends all the way down to 10Hz.)

The individual stages within the amplifier are also directly coupled with a transformerless complementary-symmetry driver stage, and Darlington type capacitorless coupling between the voltage amplifier stages.

As a result, in part, of this unique approach, the TA-3200F produces 200 watts of continuous (RMS) power at 8 ohms, across the entire frequency range from 20 to 20,000 Hz; IHF Dynamic Power is rated at 320 watts into 8 ohms (and fully 500 watts into a 4-ohm load).

But more important by far is the quality of the sound; intermodulation and harmonic distortion levels are held to a mere 0.1% at full rated output, and 0.03% at the more likely listening level of one-half watt. The signal-to-noise ratio is an incredible 110dB. And the full damping factor of 170 is maintained down to the lowest, most critical frequencies (another advantage of the capacitorless output circuit).

The companion TA-2000 preamplifier also boasts vanishingly low distortion and a wide signal-to-noise ratio, but this is less unusual in a preamplifier of the TA-2000's quality (and price) What *is* unusual is the performance of the phono and tape head preamplifier circuits; for though they have sufficient sensitivity (0.06mV) for the lowest-output cartridges (even without accessory transformers), these preamplifier circuits are virtually immune to overload—even with input signals 80 times greater than normal.

Their sole vice: they are hardly inexpensive

Of course, at a price of $329.50 for the TA-2000 preamplifier, and $349.50 for the TA-3200F power amp, this sytem cannot be considered other than a luxury, but then, it was intended to be. For there are those to whom fulfillment of prosaic tasks is unfilling. And among them are not only many of our customers, but also many of our engineers. Sony Corporation of America, 47-47 Van Dam St. Long Island City, New York 11101.

SONY®

Porsche is a trademark of Dr -Ing h c F Porsche KG

In reality several factors creep in which may make this supposition incorrect. First, belt thickness comes into picture as the belt wraps itself around the pulley. Effective radius of the pulley increases approximately by 1/3 of the thickness of the belt. Flexible belts used for such drives, as they move around the pulley, have their outer surface stretch but inner surface contract. Since drive and driven pulley are usually of different diameters (radii) and since the same belt rides over both pulleys, their effective diameters change by different percentages changing the calculated pulley ratio. This is why when a thicker belt is substituted for the original, output speed will change, not greatly but usually enough to cause trouble in drives where speed determines pitch and the correct duration of playback time. (Correct speed is important for musicians because of pitch and to broadcasters for programming).

A couple of other factors also must be considered: belt slippage and loading effect. Each is interdependent. Speed of the turntable is adjusted to be a little faster without the cartridge touching the grooves so that when a record is played, the speed will be correct.

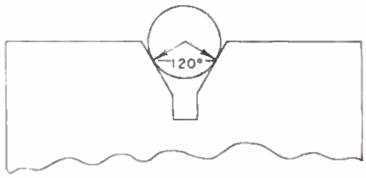


Figure 3. This groove in an O-ring round belt will minimize thickness effect.

The difference in this case is a very small part of 1 per cent of nominal speed. For the home entertainment field it can be safely disregarded. Some drive systems have provision for tension adjustment of the belt. As the belt is stretched its thickness is decreased therefore, effective diameter of the pulley changed, and so the speed.

Drives using precision-ground flat belts are more acurate than ones using molded O Rings. Since variations in belt thickness cause speed changes, proper design of the V groove on the pulleys can minimize the effect of these variations. Increasing the fly-wheel effect by making the pulleys heavier and larger, and belts softer and longer will accomplish the same effect. However, fast starts can not be achieved using heavy flywheels.

There is a new breed of belts which assures synchronous speed. These belts are in a form of round or flat belt with periodic protrusions on the side. These belts require special precision and costly pulleys and they can not operate as quietly as plain flat or round belts. However, these belts can offer fast starts and accurate average speed. Flutter also can be a problem with synchronous belts. In one of the popular transcription-type turntables such belts were used between the motor and vertical worm gear system. In order to isolate the vibration, noise, and flutter generated by the belts, the cabinet had to be internally acoustically treated and a soft coupler installed between the turntable drive shaft and the drive system.

More complicated but better performing are those systems where speed of the output shaft is electronically sensed, then impulses directed to the motor control circuit thereby keeping the speed constant. This way a closed loop servo-drive system can accomplish excellent speed control, fast starts, instantaneous speed change, independence from the load and compactness. This sort of a system with the tone wheel on the output shaft and a transistorized motor drive system has found application in small professional tape recorders. It seems that this approach of using a tone wheel to control converter frequency or current for motor drive will be the system of the future for a good deal of professional equipment. This approach will also allow miniaturization of the mechanical drive, eliminate heavy flywheels and reduce power consumption. ■

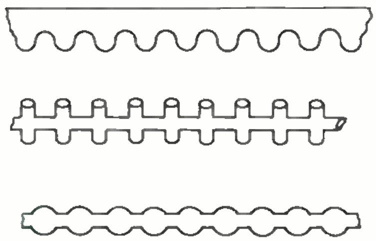


Figure 4. Various synchronous belt shapes.

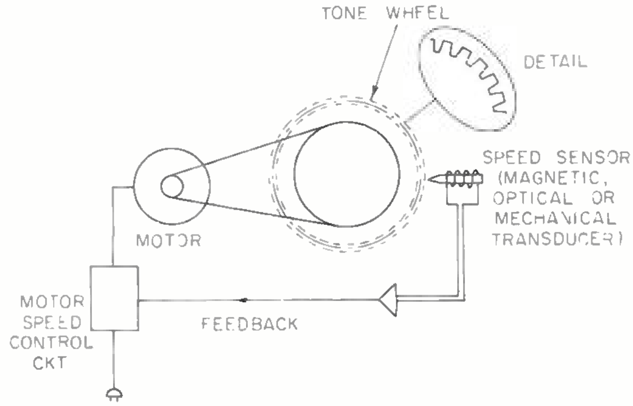


Figure 5. This circuit represents closed-loop speed control.

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.

You don't have to be an experienced writer to be published. But 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.

NEW PRO-4AA

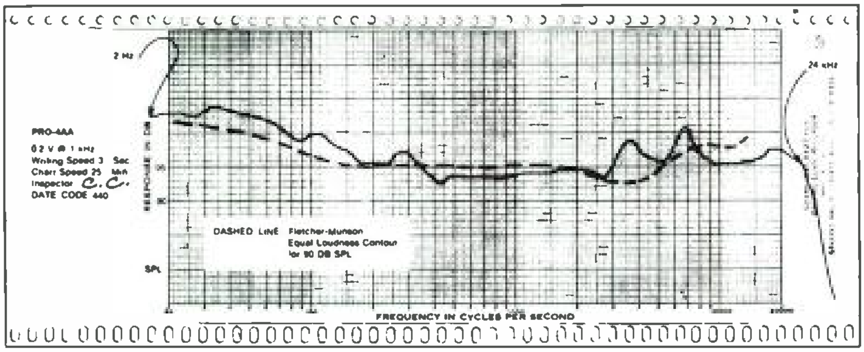
The good sound you monitor is the Sound of KOSS

Better than speakers

Here is a new, rugged, wide-range dynamic headset with good sensitivity that delivers precision monitoring far beyond that obtained by speakers. You exclude outside noise; you eliminate disturbing room reflections. Freedom from distortion and flatness of response are exceeded only by the famous Koss electrostatic units. You'll find that the "Sound of Koss" is the *good* sound.

Price $60.00

TYPICAL RESPONSE CURVE OF PRO-4AA



WHAT THE SPECS DON'T TELL YOU—

EXTENDED BASS RESPONSE TO 2 CYCLES/SECOND—The compliance of the diaphragm is large, the seal is good against the ear and the physical design permits very long excursions to give you lo-lo bass at full level—and the PRO-4AA does it with liquid-filled cushions for perfect seal and superlative comfort, whether you wear glasses or not. This bass range response *exceeds by 4 octaves* that needed to insure good low-end monitoring.

TAILORED CURVE SHAPE FOR FATIGUE-FREE LISTENING — Characteristically the PRO-4AA runs ± 3 dB except below 150 Hz and for a rise at 6.5 kHz. As long as we must have *some* variation, we put the belly of the curve in the mid-frequencies to follow closely the Fletcher-Munson equal loudness contour for 90 db SPL at 1 kHz. This keeps fatigue low in long, intense sessions because you don't reach for the extremes of the range as you tire and your ears become less sensitive. Last, but not least, the music sounds *good*.

HIGHER HIGHS THAN YOU CAN USE—The 24 kHz point on the chart is about where the "2" is in the figure 1521—the PRO-4AA needs longer chart paper than this standard charting equipment provides. Even though you can't hear these frequencies, the capability promotes good transient response in the range you *do* hear.

WHY DOES KOSS RAISE THE BOTTOM?—The wide-range coupler used for Koss measurements effects a perfect seal to promote high bass level almost to dc. An air leak to the sealed cavity lowers the bass response level. Koss feels that modern side-burns and luxuriant hair are good for a full 1-second leak or 4 dB less. This makes the PRO-4AA the flattest dynamic headset we know how to design at this time!

See and hear the PRO-4AA at your dealer today or write the factory for a 16-page catalog on "How to Choose Stereophones."

ELECTRICAL SPECIFICATIONS

Frequency Response Range, Typical: 10-20,000 Hz ■ **Efficiency:** Medium ■ **Total Harmonic Distortion:** Negligible at 95 dB SPL. ■ **Source Impedance:** Designed to work direct from 4-16 ohm amplifier outputs. When used with headphone jacks where series resistors are employed, response should not be measurably affected, but slightly higher volume settings will be required. ■ **Power Handling Capability:** Maximum continuous program material should not exceed 5 volts as read by an ac VTVM (Ballantine meter 310B or equal) with average indicating circuitry and rms calibrated scale; provides for transient peaks 14 dB beyond the continuous level of 5 volts.

PHYSICAL SPECIFICATIONS

Cushions: Fluid-filled for high ambient noise isolation averaging 40 dB throughout the audible range. ■ **Headband:** Extendable, stainless steel with self-adjusting, pivoting yokes; conforms to any head size. ■ **Boom Mount for Microphone:** Knurled, anodized, aluminum knob on left cup with threaded shaft and 2 compressible rubber washers; accepts all standard booms. ■ **Headset Cable:** Flexible, 4 conductor coiled cord, 3 feet coiled, 10 feet extended. ■ **Plug:** Standard tip, ring and sleeve phone plug. ■ **Element:** One inch voice coil virtually "blow-out proof"; takes surges up to 20 times rated maximum power levels. Has 4 square inches of radiating area from 2 mil thick mylar diaphragm. ■ **Weight of Headset Only:** 19 ounces.

KOSS Stereophones

Koss Electronics Inc., 4129 N. Port Washington Rd. Milwaukee, Wis. 53212 • Koss Electronics S.r.l. Via Vallorta 21 20127 Milan, Italy

Circle 18 on Reader Service Card

ARNOLD SCHWARTZ

THE FEEDBACK LOOP

• Time constants can be a powerful tool for simplifying circuit analysis. It has been my experience that by using time constants in designing and analyzing R-C circuits one can virtually dispense with calculating impedance. Although the use of time constants can be as exact as any other approach, if we simplify some of the numbers a system of approximations (all we really need to solve most problems) can be worked out so that most R-C circuit calculations can be performed very simply. By using time constants and a little memorization you can greatly increase your speed in analyzing and designing R-C and R-L circuits.

Time constants refer to circuits having a resistance and capacitance, or resistance and inductance. *Figure 1* shows two circuits we are all familiar with. In (A) a voltage generator is driving a series circuit containing R and C; the output voltage is taken across C. In (B) a voltage generator is driving a series circuit containing R and L; the voltage output is taken across R. The response of both circuits is the same. At low frequencies the output is flat (see *Figure 2*). At frequency *F* the reactance equals the resistance and the response is down 3 dB. This is called the knee of the curve. One octave below the knee ($\frac{f}{2}$) the response is 1 dB-down, and on decade below the knee ($\frac{f}{10}$) the response is 0.1-dB down. One octave above the knee (2F) the response is 7-dB down, and the curve will gradually approach 6-dB-per-octave as frequency increases.

The same circuits are shown in *Figures 3*(A) and *3*(B) but with the output taken across the resistance in the R-C circuit, and across the inductance in the R-L circuit. The response of these two circuits is the same. At high frequencies the output is flat (see *Figure 4*). At frequency F the reactance is equal to the resistance and the response is down 3 dB. One octave above the knee the response is 1 dB down, and one decade above the knee the response is 0.1 dB down. One octave below the knee the response is 7 dB down, and the curve will approach 6-dB-per-octave as frequency decreases.

TIME CONSTANTS

The time constant of the R-C circuit of *Figure 1*(A) and *3*(A) can be derived by equating the resistance to the capacitive reactance:

$$R = X_c = \frac{1}{2\pi FC}$$

Where X_c = capacitive reactance
C = capacity
F = frequency where $X_c = R$
T = time constant

If we solve this equation for RC we get:

$$RC = \frac{1}{2\pi F} = T$$

That is we define $\frac{1}{2\pi F}$ as the time constant (T) of the circuit. In

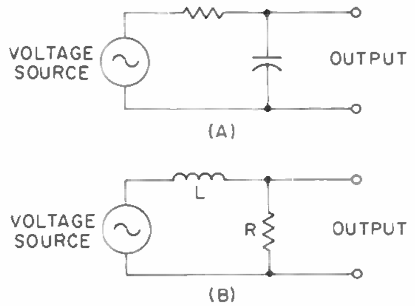


Figure 1. At (A) an R-C circuit; at (B) an R-L.

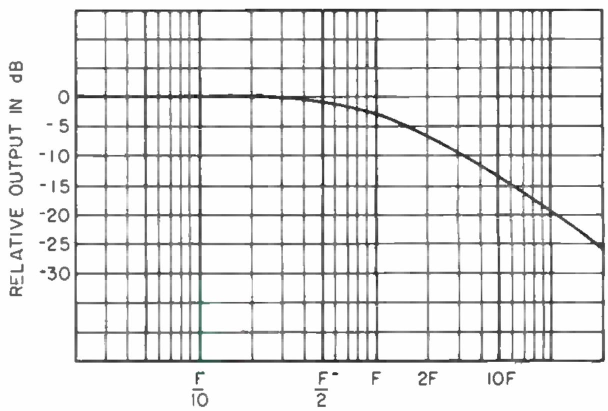
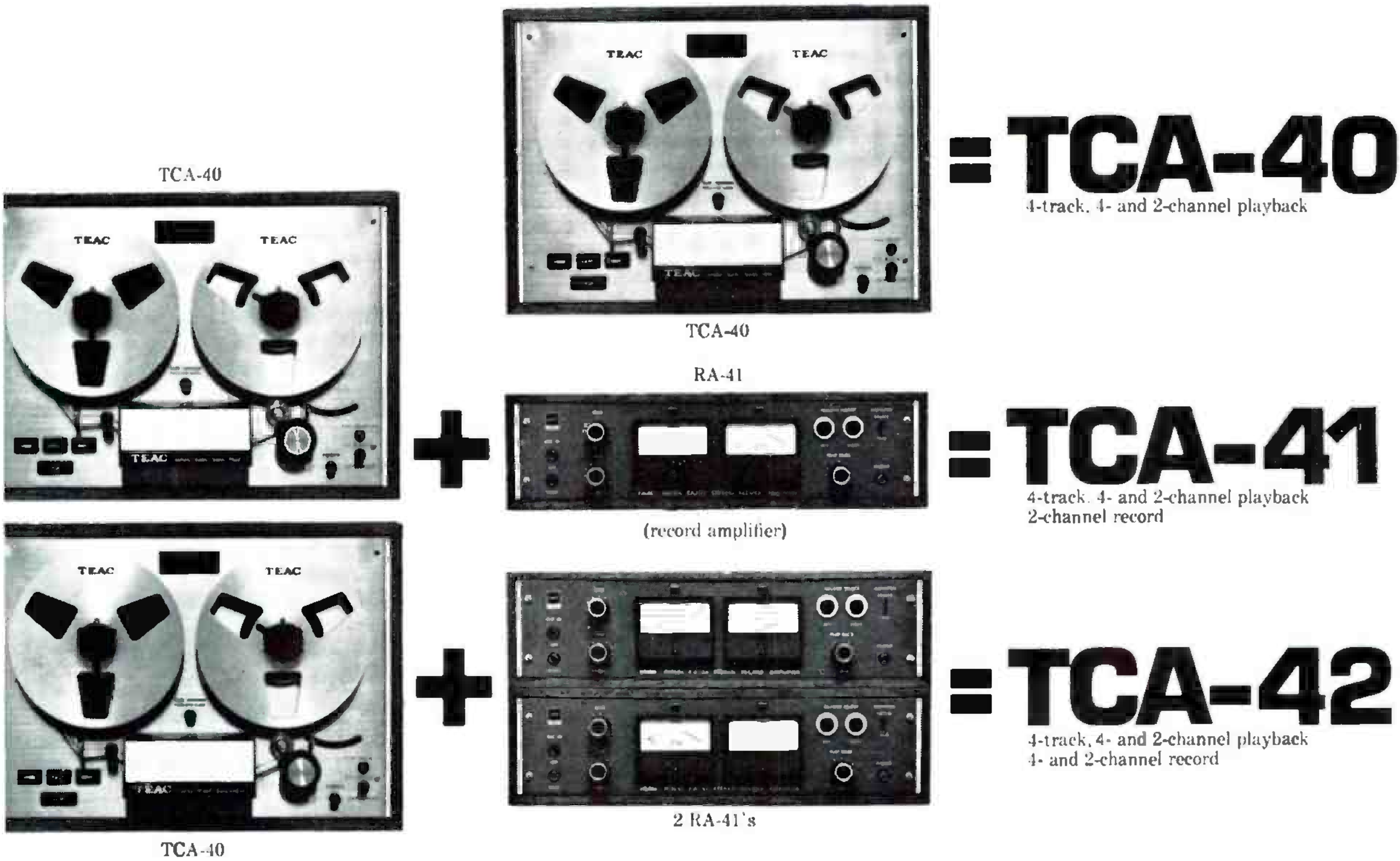
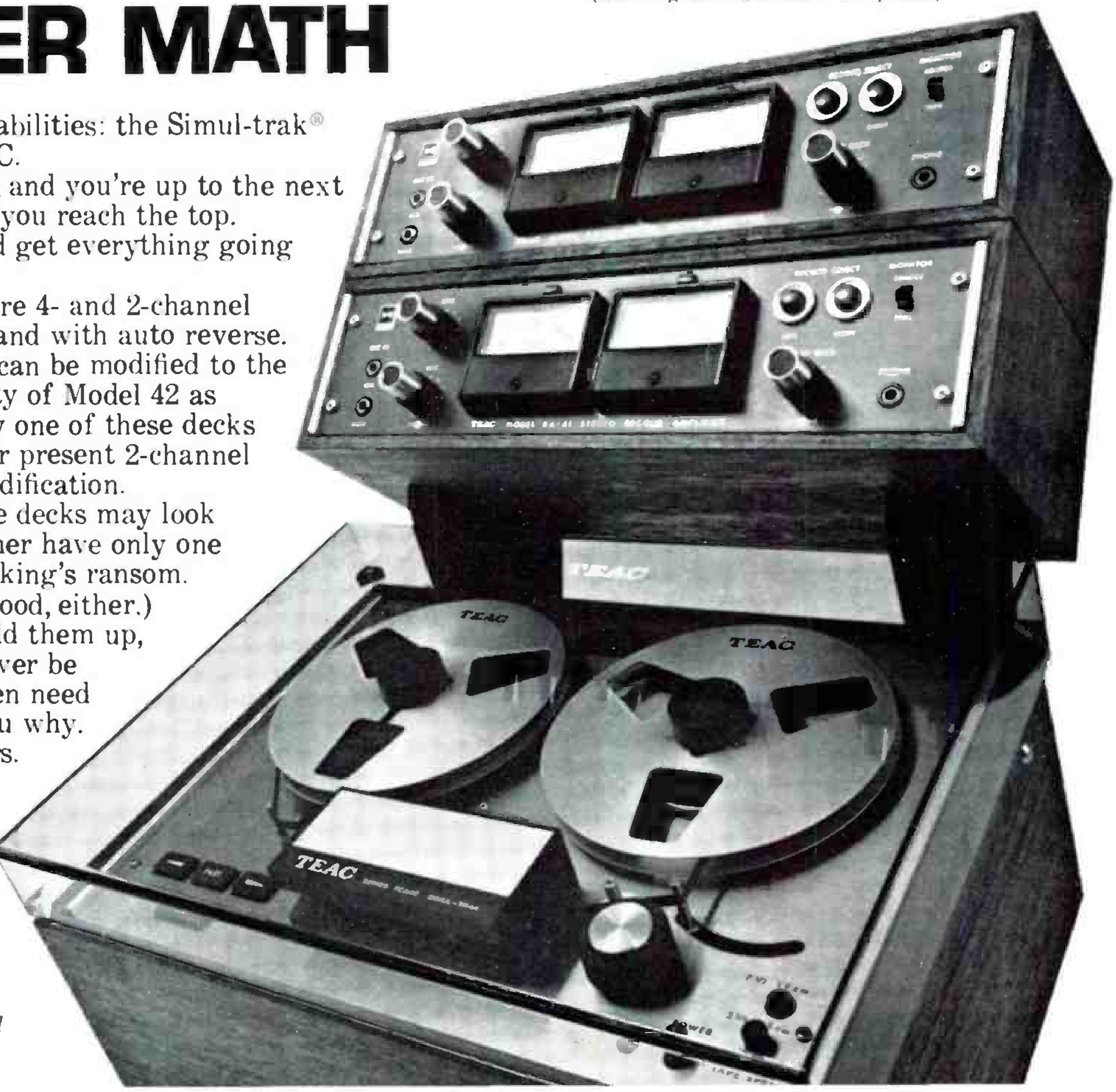


Figure 2. The response of the R-C and R-L circuits shown in Figure 1.



(Mounting cradle and dust cover optional)

HIGHER MATH



Three decks, three capabilities: the Simul-trak® TCA Series from TEAC.

Buy one, add onto it, and you're up to the next model. Keep going till you reach the top. Or start at the top, and get everything going for you at once.

All three units feature 4- and 2-channel playback – the only brand with auto reverse. And Models 40 and 41 can be modified to the full 4-channel capability of Model 42 as shown. Meanwhile, any one of these decks is compatible with your present 2-channel equipment without modification.

Other 4-channel tape decks may look like ours. But they either have only one motor – or they cost a king's ransom. (They don't sound as good, either.)

And any way you add them up, the TCA Series can never be obsolete. You don't even need a slide rule to show you why. Just a good pair of ears.

MEMBER INSTITUTE OF HIGH FIDELITY INC.

TEAC

:AC Corporation of America
)0 Colorado Avenue
nta Monica, California 90404

Circle 22 on Reader Service Card

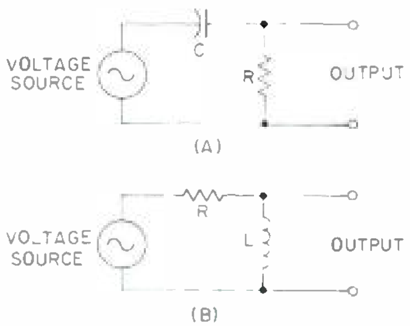


Figure 3. Another (A) R-C circuit and (B) R-L circuit.

addition, the resistance multiplied by the capacitance (R) X (C) is equal to the time constant. If the resistance is in ohms and the capacity is in microfarads then the time constant is in microseconds. In *Figure 1*(B) and *3*(B) the time constant can be derived by equating the resistance to the inductive reactance:

$$R = X_L = 2\pi F L$$

$$\frac{L}{R} = \frac{1}{2\pi F} = T$$

where L = inductance

So that $\frac{1}{2\pi F}$ is equal to the time constant, and $\frac{L}{R}$ is also equal to the time constant.

TIME CONSTANTS AND FREQUENCY

Each audio frequency has a corresponding time constant. *Figure 5* is a graph of time constant as a function of frequency, derived by using the formula $T = \frac{1}{2\pi f}$. We can spot some of the familiar time constants such as 75 microseconds at a frequency of 2,120 Hz for the f.m. pre-emphasis. The RIAA recording characteristic is defined as the combined response of three circuits having time constants of 3,180 (50 Hz), 318 (500 Hz), and 75 microseconds. The "trick" in using time constants is to commit this graph to memory so that we can incorporate the frequency-time constant relationship into everyday thinking. At first sight this looks like a formidable and unpleasant task. However, the entire time constant-frequency relationship can be boiled down to two numbers: 1 and 3. *Figure 6* shows a table of approximate time constants for frequencies of 100 Hz, 1,000 Hz, and 10,000 Hz; the time constant is the same at all these frequencies except that the decimal point is moved one place to the left as frequency increases. The same relationships are shown at frequencies of 30 Hz, 300 Hz, and 3,000 Hz. If we remember

FREQUENCY (Hz)	TIME CONSTANT (microseconds)
100	1,600
1,000	160
1,0000	16
30	5,200
300	520
3000	52

Figure 6. A table of approximate time constants.

that the time constant at 1,000 Hz is 160 microseconds, and the time constant at 300 Hz is 520 microseconds, then it is no problem to remember time constants at frequencies one tenth and ten times these two frequencies. How about frequencies that don't fall into our convenient 1 and 3 pattern? What is the time constant at 500 Hz? The time constant increases as frequency decreases, and since we know the time constant at 1,000 Hz (160 microseconds), the time constant at 500 Hz will be double, or 320 microseconds. (We now know the time constant at 50 and 5,000 Hz as well) The time constant at 2,000 Hz is half that at 1,000 Hz, or 80 microseconds. (We now know the time constants at 20 Hz, 200 Hz, and 20,000 Hz as well.) If we multiply and divide by four we can find time constants at 25 250, 2500, 40, 400, and 4,000 Hz.

By using the same process with 300 Hz and 520 microseconds we can easily identify a number of other time constants. If we have to know the time constant at a frequency that is not some fraction or multiple of 1 and 3 then a little estimating will get us close enough. What is the time constant at 7,000 Hz? If we find the time constants at 6,000 Hz and 8,000 Hz and take the average of the two values we come up with a good approximation:

Time constant at
6,000 Hz = 26 microseconds.

Time constant at
8,000 Hz = 20 microseconds.

The average of 20 and 26 is 23 microseconds. The exact value is 22.7 microseconds, and we are close enough for all practical purposes.

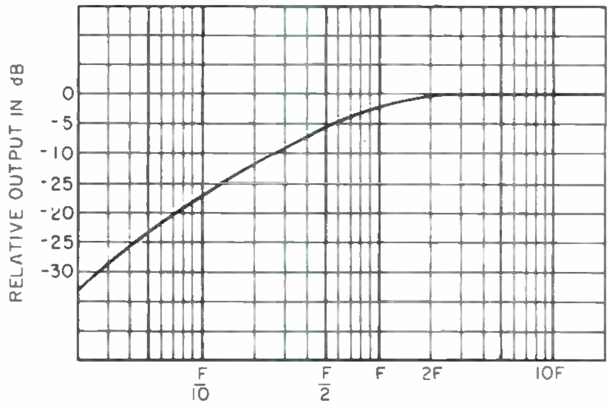


Figure 4. The response of the R-C and R-L circuits shown in Figure 3.

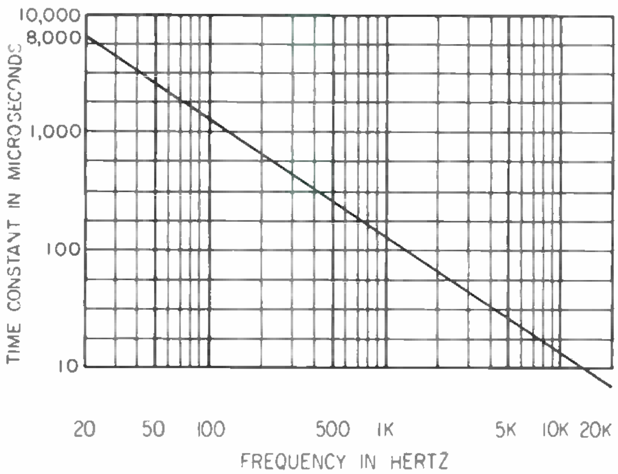


Figure 5. Time constant as a function of frequency.

Application of Time Constant

Here's an actual problem that can be solved by the use of time constants. An amplifier with a 600-ohm output impedance is driving a long audio line terminated with a 600-ohm resistor. The cable capacity is 60 micro-microFarads per foot and the length of the line is 1,200 feet. The response out to 15,000 Hz must be within ±1.0 dB. Question: will the cable capacity cause any frequency response problems?

We'll continue this topic next month and show how this problem and others like it can be easily solved by the use of time constants. ■

IF YOU DO ANYTHING WITH 1/4" TAPE, YOU CAN DO IT BETTER WITH REVOX

Automated broadcast operations
Scientific analysis
On location mastering
Tone or time changing
Audio tape quality control
Electronic music synthesis
Noise analysis
Film synchronization
Radio telescopy
Language laboratory
Machine tool control
Phonetic analysis
Radio telemetry
Industrial research
Information retrieval
Electrocardiography
Making calibration tapes
Tape mastering with SELFSYNC
Data storage from digital computers

And that's a simple statement of fact.

From the moment it was introduced, the Revox A77 was hailed as a recording instrument of unique quality and outstanding performance. The magazines were unanimous in their praise. Stereo Review summed it all up by saying, "We have never seen a recorder that could match the performance of the Revox A77 in all respects, and very few that even come close."

So much for critical opinion.

Of equal significance, is the fact that the Revox A77 rapidly found its way into many professional recording studios.

But what really fascinates us, is that the A77 has been singled out to perform some unusual and highly prestigious jobs in government and industry. The kinds of jobs that require a high order of accuracy and extreme reliability.

Take NATO (the North Atlantic Treaty Organization) for example. When they wanted a machine to standardize on, a machine that would lend itself to use in a wide variety of circumstances. And most importantly, a machine that was simple to use, the logical choice was the Revox A77.

Or take the governmental agency that wanted an unfailingly reliable tape machine to register and record satellite bleeps. The choice? Revox.

Or the medical centers that use specially adapted A77's for electrocardiographic recording.

We could go on and on (see accompanying list), but by now you probably get the point.

No other ¼" tape machine combines the multi-functioned practicability, unfailing reliability, and outstanding performance of a Revox.

If you have a special recording problem that involves the use of ¼" tape, write to us. We'll be happy to help you with it.

And if all you want is the best and most versatile recorder for home use, we'll be glad to tell you more about that too.

REVOX DELIVERS WHAT ALL THE REST ONLY PROMISE.

Revox Corporation
212 Mineola Avenue, Roslyn Heights, N.Y. 11577
1721 N. Highland Ave., Hollywood. Calif. 90028
In Canada: Tri-Tel Associates, Ltd., Toronto, Canada

Circle 23 on Reader Service Card

John M. Woram

THE SYNC TRACK

• Before continuing with a discussion of the correlation meter, it might be well to digress a bit into—of all things—logic! Aristotelian logic in fact. For those who don't see the immediate connection, Aristotle's philosophy recognizes that a statement may be either true or false, but never *partially* true or false.

In addition to that bit of heavy thinking, Aristotle did a lot of writing, and died in 322 B.C. Then, in the 1840's the English mathematician George Boole, presumably having nothing better to do with his time, developed a system for presenting the logic of Aristotle in a mathematical notation. Although his mother was no doubt very proud, nobody else much gave a damn until recently when it was realized that old George had come up with a very handy way of describing the state of a switch, or any other two condition device.

In Boolean algebra, a statement that is true may be represented by a *1*; a statement that is not true may be represented by a *0*. In modern circuit design, a switch that is closed may be represented by a *1*, and the same switch, when open, may be represented by a *0*. Likewise, any conducting path may be a *1*; a non-conducting path a *0*.

In *Figure 1*, each circuit (A) and (B) contains three switches, w, x, and y. Depending on their positions—on or off—(*1* or *0*) there either is (*1*) or is not (*0*) an output at point z.

If you can stick this out a little longer, you'll see the point of getting so involved in defining what seems to be a simple collection of on/off switches. Applying Boolean algebra to *Figure 1*(A), we can say that there is an output *1* at point z if either w *or* x *or* y are closed (*1*). In *Figure 1*(B), z = *1* if, and only if, w *and* x *and* y are all closed.

THE TRUTH TABLE

A truth table is a tabulation of all possible combinations, together with the result—in this case, an output, or the lack of one. Since each switch has two possible states, and there are three switches, there are 2^3 = 8 conditions which may be described. *Figure 2* is the truth table for these 8 conditions.

The circuit in *Figure 1*(A) may be referred to as an *or* circuit, since there will be an output if w *or* x *or* y are closed. *Figure 1*(B) is an *and* circuit, since there will be an output only if w *and* x *and* y are all closed. The *or* function is often represented by a + sign and the *and* function by a ·, however these symbols should not be confused with their traditional values of *plus* and *times*, respectively. Perhaps it would have been less puzzling to use some new symbols, but this way anyone who writes 1 + 1 = 2 is immediately exposed as an outsider, and all the in crowd can then have a big chuckle.

BINARY ARITHMETIC

Now that we have a system that uses only 2 digits (*0* and *1*), think of the possibilities of a numbering system that uses only these digits. If we could represent any number in our decimal system by some combination of *0*'s and *1*'s, rather than using the usual 0 thru 9, then any number could be created by a collection of switches (or lights perhaps). Each switch or light simply would be either on (*1*) or off (*0*). In order to register all the numbers in the decimal system, the same device would have to be capable of indicating ten different conditions, making it quite complex, and probably difficult to interpret quickly.

To see how this binary system works, let's consider how we construct a numerical series in our regular decimal notation. Very simply, we start with a 0, count up to 9, and then begin all over again, adding a 1 in the space to the left, as shown in column (A) of *Figure 3*. In the binary system, we also begin at 0, but only go as far as 1 before starting again. A binary series is shown in column (B). Just for no reason, I've included a tertiary notation in column (C) for further comparison. For any row, the number in each column represents the same physical quantity—the only difference being the system in which it is expressed.

Obviously, it takes more spaces to indicate any number in the binary system, yet the simplicity of operation of a two-state device more than compensates for this. And, it's remarkable how much sophisticated work may be done by devices that indicate—or otherwise react to—the presence (*1*) or absence (*0*) of information. For example, consider the compatability, or correlation, of two signals that make up a stereo program. We want to devise an indicator that will be insensitive to frequency response, musical content, dynamic range and all the other complex variables that do not concern

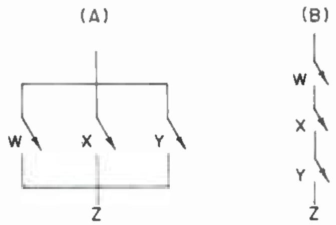


Figure 1. Two simple switching circuits.

us at the moment. The indicator must tell us *only* what will happen when a stereo program is played back (or broadcast) in mono, or, when a stereo disc is cut. Of course, the disc still yields a stereophonic program, but it does so from a single groove, which contains both vertical and lateral information. As we know, out of phase information produces vertical motion of the cutting stylus. Actually, the term *out of phase* may be a little misleading when measuring two complex wave forms. We should rather speak of the amount of correlation between them. For example, look at the two complex waves shown in *Figure 4*. There's not much that can be said about the phase relationship between them, but what can be measured are the intervals during which the waves are alike, or unlike, in sign. Now, if we can produce a rectangular wave that is positive when the waves are alike (+, + or −,−), negative when they are unlike (−,+ or +,−), and zero in the absence of information of one (or both) waves, the average value of this wave will give us an indication of the amount of time during which the waves are correlated (alike in sign) or uncorrelated (unlike in sign).

Since each wave has only three possible relevant conditions (positive, off, negative) and there are two waves, we have 3^2 = 9 possible conditions, and the rectangular wave to be produced must likewise have the same three possible conditions. The truth table in *Figure 5* shows all possible conditions together with the required condition for our rectangular wave. Notice that this truth table discards all the irrelevant information (frequency, dynamic range, level, *et. al.*) and answers only one question. Are the waveforms L and R correlated, or are they not? By carefully examing the truth table, we can get an idea of the circuit required to produce our rectangular wave. By inspection, we need a device that will produce a positive voltage in the presence of like signs (conditions 1 and 9) a negative

CONDITION OF SWITCHES			OUTPUT	
w	x	y	Za	Zb
0	0	0	0	0
0	0	1	1	0
0	1	0	1	0
0	1	1	1	0
1	0	0	1	0
1	0	1	1	0
1	1	0	1	0
1	1	1	1	1

Figure 2. A truth table for the switching circuits of Figure 1. Za=output condition for Figure 1(A); Zb=output condition for Figure 1(B).

voltage in the presence of unlike signs (conditions 3 and 7) and will shut off when either, or both, waveforms are zero (conditions 2,4,5,6, and 8).

Figure 6 shows the first step in designing such a circuit. The black boxes labelled either *and gate* or *or gate* are transistorized versions of the switches shown earlier in *Figure 1*. *And gate 1* will conduct only if there is a potential at both points a and c, a condition which will exist only if diodes a and c are both conducting. And this will only happen if both waveforms L and R are negative. If they are both positive, diodes b and d will conduct, and *and gate 2* will conduct. In either case, *or gate 1* will conduct and a positive voltage will be found at point x. *And gates 3* and *4* will conduct only when waveforms L and R are unlike in sign, therefore *or gate 2* will conduct, and a negative voltage will be produced at point y. In the absence of information from either waveforms L or R, neither of the associated diodes will conduct, consequently none of the *and gates* will turn on, and there will be zero potential at both points x and y. Points x and y may be connected together, filtered, and fed to a zero center meter. A zero reading will indicate that during the measurement interval the waves L and

Figure 3 A comparison of decimal, binary, and tertiary notations.

	COLUMNS		
	A DECIMAL	B BINARY	C TERTIARY
ROW	0	0	0
	1	1	1
	2	10	2
	3	11	10
	4	100	11
	5	101	12
	6	110	100
	7	111	101
	8	1000	102
	9	1001	110
	10	1010	111
	11	1011	112
	12	1100	120

R are alike and unlike for equal amounts of time. Readings to the right of zero indicate that the waves are more alike than unlike. Full positive deflection would indicate the waves were identical (for example, a mono signal). Readings to the left of zero would indicate the opposite condition, and a full negative deflection would indicate that L and R were mirror images of each other. (For example, two sine waves of the same frequency, and 180 degrees out of phase.)

Before too long, I hope to have the actual circuit values worked out, and for all I know, it might even work. If and when it does, I'll get back to you.■

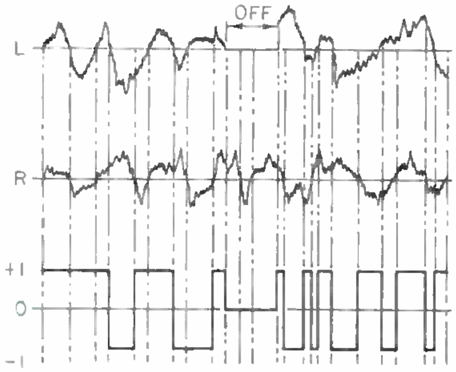


Figure 4. Two complex waves, and a rectangular wave derived from them.

Figure 5. A truth table for the waveforms shown in Figure 4.

CONDITION	L	R	RECTANGULAR
1	-	-	+
2	-	0	0
3	-	+	-
4	0	-	0
5	0	0	0
6	0	+	0
7	+	-	-
8	+	0	0
9	+	+	+

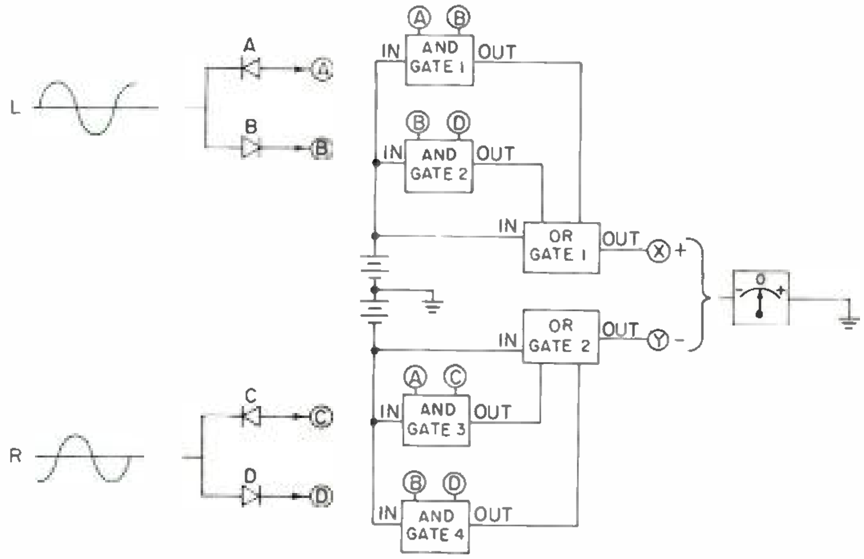


Figure 6. A block diagram for a correlation meter.

NORMAN H. CROWHURST

THEORY AND PRACTICE

• Here is another good question for this column. And one that has been repeated several times already. How can the same job done by a hybrid coil (or more accurately, two of them) be achieved without any transformers?

The first step here is to see how and why hybrid coils work, and what is required for them to work correctly. Then we can design a non-transformer circuit to do the same thing. *Figure 1* shows the classic hybrid coil circuit, used all over the place on long telephone lines. It amplifies signals going either way, without getting them mixed up.

It is very important for them not to get mixed up at these amplifier points, because if one amplifier's output fed the other's input, and *vice versa*, it would be a terrific feedback loop, and nothing else could happen. The hybrid coil's job is to see that signal coming from the line on the left goes only to the amplifier (which is not so important, because the other amplifier's output will not receive signal as input) input designed to receive it.

Then the output of that amplifier must feed only the line on the right. It must not feed, at all (which means signal should be far lower than maximum amplifier gain) into the other amplifier's input, which signals coming *from* the line on the right do.

The hybrid coil creates a sort of signal one-way street.

An important part of its operation is a dummy line, which is an impedance made up to match that presented by the line. Thus signal coming from an amplifier output energizes the core of the hybrid coil via its own winding. The other two windings, between them, feed the real line and the dummy line. If these two impedances match, the center taps of the windings to which they are connected will be null points, so no input goes to the other amplifier.

Signal coming in, on the other hand, goes through the two windings into the dummy line load. In this instance, the impedance of the amplifier output comes into it, because this reflects as a load in series with the dummy line, through a 2:1 step-down. The two center-tapped coils are in series, in phase, so that a perfect transformer with no load connected to the third winding would appear open circuit.

The amplifier impedance allows current to flow through to the dummy load, and the center tap points are one each nearer to each line input point, along the voltage division that runs consecutively from top line, through the first coil to the dummy load, through the second coil to the lower line.

Thus, signal gets to the amplifier input when it originates from the line, but not when it originates from the other amplifier.

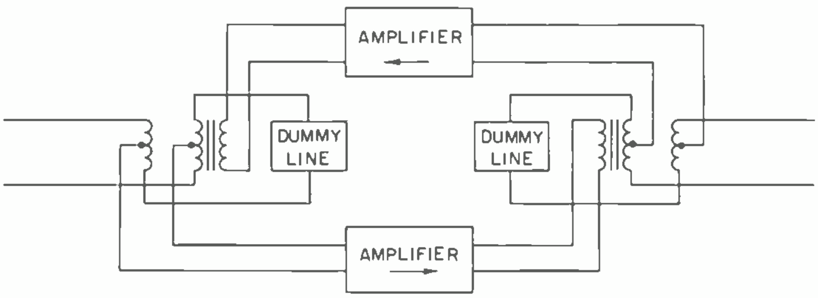
An important feature of this circuit is that, not only do the impedances of the real line and the dummy line have to match, but all circuits are balanced. Amplifiers with balanced inputs and outputs are used. Note that there is a difference between *balanced* and *floating*, which for some respects is similar, in that neither situation has either side grounded.

A balanced circuit has equal and opposite voltages on its two legs at all times. A floating circuit may have different voltages on its two legs.

Figure 2 shows a way of eliminating the transformers, provided that losing balance does not matter. The central feature is a bridge, consisting of the line, a dummy line and two equal resistors. The amplifier inputs and outputs are connected to opposite corners of the bridge.

The amplifier output feeds both the line and the dummy line, each through one of the equal resistors, so that the

Figure 1. The classic hybrid circuit, used for inserting repeater amplifiers in a telephone line, so each amplifier handles only the signal traveling in the direction assigned to it.



points where the equal resistors join with the two line impedances form a null, to which the other amplifier input is connected. Signal coming in feeds through the dummy line from one side of the line, to the other amplifier input.

The problem with this circuit is that the line is not basically balanced: the two sides connect to different impedances. There are several ways to overcome this. One would be to duplicate the bridge, but then the junction between the two circuits would not be ground, but another floating point. A complete double bridge is a possible solution, but look at *Figure 3*, as one way to go.

This merely consists of splitting the resistive arms into two, and putting half of it on either side of the line connections. From the viewpoint of the amplifier input, the points to which it is connected are still a null to signal coming from the other amplifier output.

In this circuit, the impedance of amplifier inputs and outputs enters into the calculation of what happens. Signal coming in travels from the line, through two pairs of resistors to the dummy load. Input to the amplifier is picked off across one pair of resistors.

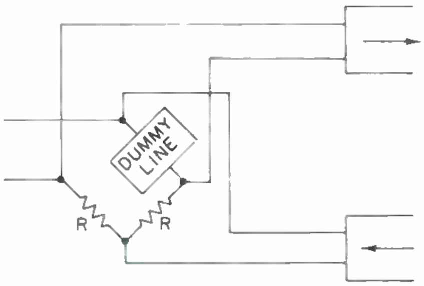


Figure 2. A simple, unbalanced arrangement that theoretically achieves the same purpose without transformers.

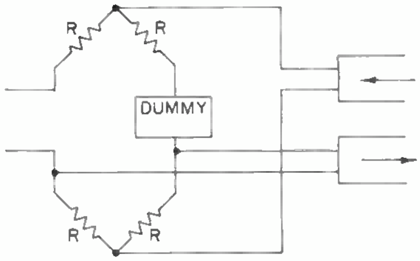


Figure 3. A slightly more involved circuit, that allows the line to be balanced, or to be adjusted so that it is.

Signal going out feeds from the amplifier output to the bridge, each pair of legs of which consist of two resistors and a line impedance, with the line impedance in the middle. The amplifier input is across one of the possible null points.

More complete balance could be achieved by using an amplifier with two floating inputs, each of which has no ground, and the signal amplified at the output would be a combination of the two signals, without taking account of the signal between the two signal inputs (across the two line impedances). Phasing would have to be correct, so that the desired input impedance added, instead of neutralizing.

An alternative is to make the amplifier input impedance high, so that it accepts a voltage input without materially loading the circuit. Line impedances are ordinarily about 500 ohms (or less). Appropriate values for R in *Figure 3* would be 250 ohms (use 240 ohms as a preferred 5 per cent value). So if amplifier input impedance is 10,000 ohms or higher, it will not materially affect the bridge.

In the transformer-type hybrid, balance over mid-range frequencies depends on the numbers of turns being precisely correct, particularly the center taps being at the centers of their windings. But balance over the entire frequency range involves being sure that the transformer behaves correctly in frequency ranges where the transfer ratio may not be solely due to the number of turns, but is also affected by their physical arrangement.

The properties that affect transformer ratios at higher frequencies are leakage inductance and winding capacitance and intercapacitance. These properties can interact to make the voltages appearing across different portions of winding quite different from the ratios determined by turns that control them at lower and medium frequencies.

Unless these three-winding transformers are carefully designed for the purpose they are to serve, the voltages across halves of a center-tapped winding can become quite unequal at higher frequencies, so the null is no longer affected correctly.

Beyond that, if the amplifier inputs and outputs are designed to be balanced, (not only in the sense of handling voltages that are equal and opposite relative to ground, but in the sense of providing equal impedance loading and source) the classical hybrid circuit works well.

Eliminating the transformers from the hybrid circuit puts more demands on amplifier design, especially if the amplifier is also not to have any transformers. The amplifier needs to handle floating inputs and to generate an output that is determined by the *difference* between signal voltage on the input terminals. If there is zero difference between the voltages, even though each terminal has a signal voltage on it, then no signal must be amplified.

One way to achieve this would be to float the whole amplifier, so its ground could have a signal difference from real ground. But this is not too feasible, because doing this at the input means the output circuit ground is also floating at the same signal voltage present on the input ground. The signal delivered to the bridge at the output end would not be a true push-pull circuit, as it should be, but would have a floating push-push component.

So it is preferable—and this is particularly important for the higher frequencies—for amplifier grounds to be true grounds, and for supply circuits all to be at a.-c. ground.

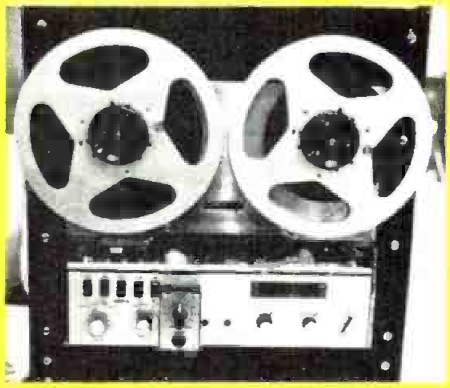
On the output end, securing a controlled, push-pull output can be achieved by using constant-current type output sources in push-pull, so the accuracy of output signal is controlled by the bridge circuit into which it feeds. There may be other ways to do this, but this is the one we shall pursue in the next issue. ■

NEW PRODUCTS AND SERVICES

STEREO ECHO MIXER

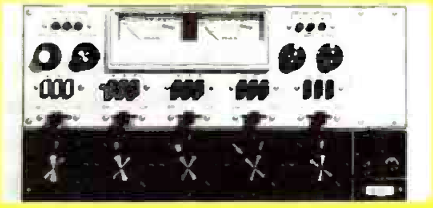
• Up to 8 inputs, 4 line and 4 mic level, are switch selectable from the front panel on this new EM-7S. Any input can be assigned to either program or echo output channel in any proportion by means of the four pan pots standard with the unit. These units are stackable allowing additional inputs to be added at any time. All mixing circuits are of the active type using integrated circuits. Output is rated at +28 dBm and mic input noise is below -127 dBm referred to the input. It fits standard racks and is 7-inches high and 8-inches deep.
Mfr: Gately Electronics
Circle 55 on Reader Service Card

SYNC RECORDER



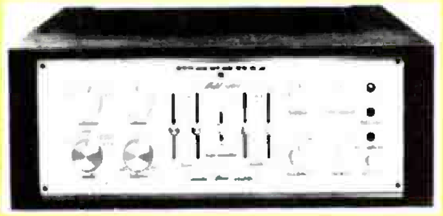
• A new version of the model A-77 tape recorder now permits separate synchronized recording of two master channels in addition to existing internally wired sound with sound, echo, etc. Called "In Sync" this version costs an additional $125.00 above the price of the basic recorder.
Mfr: ReVox Corp.
Price: $125.00 plus recorder
Circle 62 on Reader Service Card

AUDIO CONSOLE



• Fifteen inputs with two separate program outputs are featured in the model A-16R audio console. Designed for t.v. and c.a.t.v., its independent vu meters permit simultaneous program airing and production work. Other features include a removable front panel, spare controls and spare terminals for custom needs. 8 3/4-inch rack space is required, or the unit can be supplied in a cabinet.
Mfr: Sparta Electronic Corp.
Price: $995.00
Circle 62 on Reader Service Card

CONTROL AMPLIFIER



• The model 1200 contains the versatility and convenience features usually found in the most elaborate preamp-control consoles, and includes in one compact unit the ultra conservative ratings and ruggedness of separate basic power amplifiers. Total continuous power with both channels driven at or below rated distortion 20 Hz to 20,000 Hz is 200 watts into 8 ohms (100 watts/channel); 250 watts into 4 ohms; 100 watts into 16 ohms. Frequency response is ± 1.5 dB, 3Hz to 100,000 Hz and signal-to-noise ratio is 100 dB above 2 mV.
Mfr: Superscope, Inc. (Marantz)
Price: $595.00
Circle 63 on Reader Service Card

AUDIO SIGNAL DELAY SYSTEM

• The DELTA-T model 101 audio signal delay system is the world's first piece of professional audio equipment incorporating digital processing of wide band audio signals. It incorporates no mechanical moving parts, but instead uses the latest advances in MSI solid-state technology. The model 101 incorporates within a 7 inch rack unit the equivalent of some 600,000 discrete solid state devices. It provides up to five separately controllable outputs from a single input, each adjustable in 5 ms time delay steps to a maximum of 320 ms, while maintaining a s/n ratio of 60 dB and a response range from 30 - 12,000 Hz with both harmonic and intermodulation products below 1 per cent for line level outputs. The DELTA-T system, whose applications range from public address distribution to acoustic research, reverberation enhancement and electronic music uses, has balanced inputs and outputs, 20 dB of extra gain, delay by-pass switches for each output and may be powered from all of the world's power line configurations. Modular construction permits the user to buy only the number of outputs and the amount of delay time he actually needs, and to upgrade at any time in the future to the unit's full complement. A comprehensive brochure is available.
Mfr: Gotham Audio Corp.
Price: from $3,192.00
Circle 78 on Reader Service Card

WHEATSTONE BRIDGE

• A wide range of d.c. resistance testing is possible without the use of external batteries. The Norma Wheatstone Bridge, type 1802-30301 is manufactured in Vienna, Austria. It has been designed to measure resistance from 0.08 ohms to 120 megohms in nine ranges. Six push-button multipliers and single dial reading are used to select the ranges which are indicated on a mirror-scale galvanometer. Test voltages for the ohm and kilohm ranges are supplied by self-contained batteries, while a transistor rectifier circuit supplies the test voltage for the megohm range. Accuracy is claimed to be ±0.5 per cent from 0.8 ohms to 12 megohms; outside these range it is ±2 per cent.

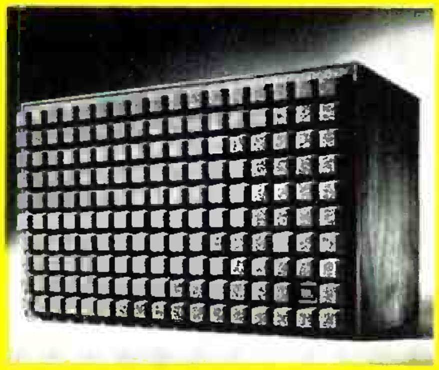
Mfr: Norma (distributed by Freed Transformer)
Price: $205.00
Circle 54 on Reader Service Card

SHOTGUN MIC ACCESSORIES

• Two new accessories increase the versatility of this company's shotgun mic line. First, is a windscreen using open cellular foam that provides reduction of input from both higher and lower frequency air turbulence. Just 20-inches long and 2.5 inches in diameter, the MZW 815 stows easily. The windscreen will not impair the frequency response nor the directional characteristics of the mic. Also shown in the pistol grip, model MZP 805. This helps take the fatigue and work out of holding a shotgun for an extended time. The hand grip can be slipped under the point of gravity and the sure lock swivel keeps the mic in a convenient angle. A threaded hole in the bottom of the handle accepts a 3/8-inch bolt for fastening to a stand or boom.

Mfr: Sennheiser
Price: Screen–$29.00; grip–$88.00
Circle 76 on Reader Service Card

BOOKSHELF SPEAKER



• The L100 Century is a decorated version of the 4310 Studio Monitor unit, and like it is contained in a volume of only 1½ cubic feet. The grille material is three-dimensional yet more acoustically transparent than cloth. Components include a 12-inch long-excursion woofer, 4-inch diameter midrange, and a small tweeter. Control exists for mid- and high-frequency driver balance.

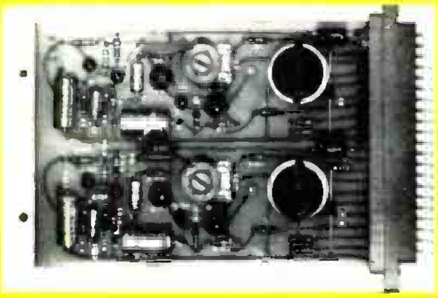
Mfr: JBL
Price: $264.00
Circle 53 on Reader Service Card

450 WATT AMPLIFIER

• A new 450-watt power amplifier system delivers continuous power into an 8-ohm load with less than 0.05 per cent i.m. distortion at any level. These high-power amplifiers were designed to provide the extra headroom necessary for tuned and voiced speaker systems in a minimum amount of space. Three 450-watt monitor amps can be mounted in a 5 1/4-inch relay rack. Peak power level for each amplifier is in excess of 1,000 watts.

Mfr: Daniel N. Flickinger & Assoc.
Circle 66 on Reader Service Card

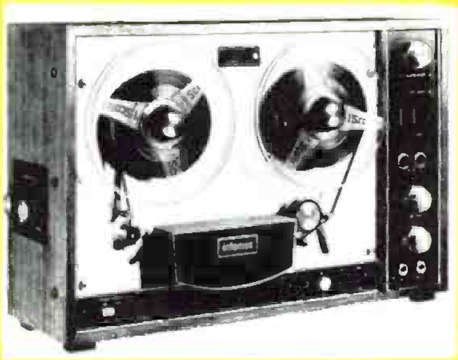
DOUBLE AUTOMATIC ATTENUATOR CARD



• This is a double automatic attenuator card–model 692DAT, an addition to the line of versatile and flexible Integra II components. Uses of the 692DAT board include gating of audio lines to eliminate unwanted low level signals (noise) or changing the gain of the lines, either by signals transmitted to same or from other sources. In addition it can also be used as a "ducker" in p.a. or specialized recording by broadcast applications. In other words, the 692DAT will handle two inputs–one input automatically reducing the gain of the other input. Unlike any other gated amplifier circuits, the auto-ten performs soft switching by using smooth and fast resistance change of CdS cells, which are actuated by fast incandescent plug-in bulbs. The new circuit offers a well-defined threshold point with resolution of 1 dB. The lowest threshold setting provides action from signals as low as −40 dBm. Two separate threshold controls are mounted on a board with provision to use external controls. Release time for the gating action can be either fixed or variable.

Mfr: Fairchild Sound Equipment Corp.
Circle 71 on Reader Service Card

EIGHT-TRACK MASTER RECORDER



• Model MR-8 is a precision recorder-reproducer for mastering or remastering 8-track music on quarter-inch tape. It is designed to produce the master tape to be used on the Infonics 8-Track duplicator, model D-8. Equally important, it is the ideal quality control instrument to play back 8-track tape duplicates produced on the D-8 to test the quality of the duplication before loading the tape into the cartridge. The MR-8 features two interlaced record heads with a total of 8 tracks, and two interlaced reproduce heads in the same format. By actuating the program selector switches on the left side of the recorder, any of the four stereo programs can be selected in either the record or the reproduce mode, independently of each other. When mastering, one would normally monitor the stereo program being recorded by switching the playback program selector to the same position as the record selector. Each stereo program is separately recorded on the MR-8. After finishing stereo program 1, the tape is rewound at high speed; then stereo program 2 is recorded, etc. It operates at either 7½ or 3-3/4 in./sec. A music tape recorded at any speed can be played back on an appropriate recorder and remastered into a 3-3/4 in./sec. tape to serve as the master for the D-8 duplicator. If it is desired to remaster a 3-3/4 in./sec. tape, both recorders can be speeded up to 7½ in./sec. in order to do the remastering in half the normal time.

Mfr: Infonics
Price: $2,495.00
Circle 79 on Reader Service Card

MINIATURE PREAMP MODULE

• A new plug-in preamplifier and line-level amplifier which occupies only 3½ by 1½ inches of panel space is now available. Designated model 512 it has separate shielded input transformers for low and high-level signals. Transformer-coupled output levels to +30 dBm allow maximum headroom while maintaining an equivalent input noise of -129 dBm. A three-position switch provides for additional mic, line, or test oscillator input. Phantom power connections are also available. A small meter works in conjunction with the continuously variable gain control on the preamp to provide optimum amplification for all types of mics and studio conditions. Bipolar power of 15-20 volts is required. Reliability is assured through the use of this company's model 2520 operational amplifier.

Mfr: Automated Processes
Circle 59 on Reader Service Card

EDUCATIONAL HEADPHONE

• These stereo phones use an audiometer-type transducer for performance stability. The driver is impervious to temperature and humidity changes and has a frequency response of 20-20,000 Hz. Impedance is 8-16 ohms. Model 325 can be used with electronic keyboard and fretted instruments and is claimed to be ideal for music-appreciation use. Immunity to abusive use is assured by the use of ABS plastic earcups. Clutch type volume controls are in each channel.

Mfr: Telex
Price: $49.95
Circle 52 on Reader Service Card

OMNI MIC

• Smooth response, light weight, slim sillhouette and resistance to shock are features of the new BK-14A omni dynamic. It is particularly recommended for outdoor as well as indoor use, and has special screening against wind and pop noise. The cartridge is replaceable and there is provision for stand mounting. Styling is non-reflecting satin nickel in a housing that is 8-inches long and 3/4-inches in diameter. A swivel mount and 30-foot cable cand connector is supplied.

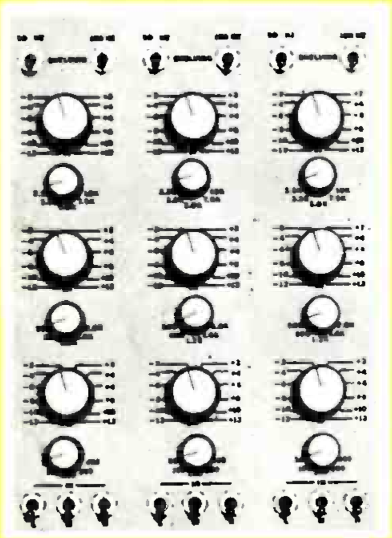
Mfr: RCA
Circle 51 on Reader Service Card

AUTOMATIC REVERSE PLAY DECKS

• The A-1250 incorporates a dual bias switch which provides a dual range bias current control which offers a wider choice of tape and maximum frequency response, with impressive signal-to-noise ratio performance. It also incorporates the TEAC symmetrical control system, and is solenoid operated. The newly developed pause control system allows fast start and stop operation in either playback or record mode, and eliminates troublesome clicks. There is also facility for easy adjustments of turntable height which compensates for variations in reel dimensions and protects valuable tape. Live monitoring and built-in microphone and line mixer are also significant features. The hyperbolic heads are mounted on a special plate assembly which assures absolute freedom from movement, and virtually eliminates head mis-alignment. A special composite head, which includes the erase and record functions. Three motors are used.

Mfr: TEAC
Price: $449.50
Circle 75 on Reader Service Card

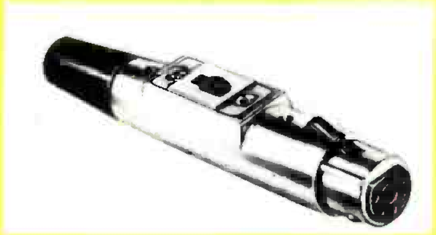
THREE RANGE EQUALIZER



• A new microphone/program equalizer, the model 502, provides simultaneous selection of equalization for low, mid, and high frequencies with a separate in/out switch for each range located for easy access. Five separate frequencies are available in each range. Low: 50Hz, 100Hz, 200Hz, 300Hz, 400Hz. Mid: 500Hz, 800Hz, 1200Hz, 1600Hz, 2000Hz. High: 2500Hz, 3500Hz, 5000Hz, 7500Hz, 10,000Hz. Reciprocal Gaussian curves (graphic type) for each frequency are selectable in 2 dB steps for a total of 12 dB of boost or attenuation. Low and high frequency shelving curves, adjustable in 2 dB steps, are also provided. Utilized in the feedback loop of this company's model 101 audio amplifier, this equalizer has zero insertion loss and is less than 1½-inches wide x 6½-inches high by 2 7/8-inches deep. Distortion is unmeasureable (less than 1/100th of 1 per cent measurement residual) under any condition of boost or cut.

Mfr: Spectra Sonics
Price: $240.00
Circle 64 on Reader Service Card

SWITCHABLE CONNECTORS



• A new line of audio connectors containing an on-off switch and designed for use with professional microphones has been introduced. The connectors, called T(*)F and T(*)FL Audio Connectors, were designed in conjunction with a leading microphone manufacturer to give professional performers full control of their microphones. Their size is compatible with that of professional microphones, and strain relief is excellent. The built-in dpdt slide switch is located so that a performer can easily find and operate it with his thumb. It has a positive detent so that it cannot be operated accidently. They are available with 3-, 4-, and 5-pin female straight cord plugs that mate with all "Q-G" male plugs and with microphones having similar insert arrangements and identical number of contacts.

Mfr: Switchcraft
Circle 65 on Reader Service Card

ELECTRONIC TIMER

• Two consecutive time periods which are easily adjustable and absolutely repeatable are provided by the model 620 programmer. Single units can perform all dealy, interval, and repeat cycles timing from 00.0 to 99.9 seconds for both periods in 0.1 second steps. Three units can be mounted in a 7-inch rack space and wired to provide interrelated event timing.

Mfr: Hill Enterprises
Price: $299.00
Circle 60 on Reader Service Card

MELVIN C. SPRINKLE

Acoustics for Audio Men, Part 1

This begins a series of articles designed to give the practicing audio professional a better understanding of the acoustic conditions under which his product may ultimately be evaluated.

WHY all the noise about acoustics? Well, the leaders in the science of the generation of sound from electrical signals (that's audio by another name) have become aware that their responsibility does not terminate in a resistor tied across the output of an amplifier (that's the way most systems are tested), nor at the loudspeaker. They have come to realize the importance of the listener being furnished with not only quality sound but also that he be enabled to hear it *properly*. The actual line of demarcation between the science of acoustics and that of audio engineering (in the electrical sense) is very thin and scarcely definable. Unless an audio engineer is going to confine himself and his activities to strictly circuit design of amplifiers and electronic equipment, he needs to know at least the rudiments of acoustics. For the man concerned with system design (such as sound-reinforcement systems or music-reproducing systems) or with transducer design (microphones and loudspeakers) a knowledge of acoustics and sound is absolutely essential. Far too many sound-reinforcing systems have been "designed," installed, and then failed to perform properly and give adequate coverage simply because the "designer" was ignorant of the elements of acoustics. Then such a system is ripped out (at the owner's expense of course) and another brand of equipment installed in a futile attempt to correct a situation which was *not* the fault of the original equipment itself, but because it was not designed and installed so as to *use* instead of violate the natural and inexorable laws of acoustics.

It is the purpose of this series of articles to present in as simple a form as possible, the rudiments (at least) of acoustics and sound transmission. We are going to begin by considering a few of the fundamentals of sound as might be taught in a college physics class. Then we are going to consider how sound travels and is transmitted, considering both the indoor and outdoor cases. We are going to consider the acoustical properties of rooms, and maybe dispel a few old wives' tales to boot. Finally we will attempt to wrap up the subject by noting some of the applications of these principles to the practical design and installation of sound systems. We are not attempting to make this a short course in how to be an acoustical engineer. But we are going to present enough material so that a sound-system designer can use his knowledge of acoustics so that his installation will work, and work well.

Melvin C. Sprinkle is a registered professional engineer and has previously contributed to db.

This is, perhaps the distinguishing mark of the professional. The amateur often cuts and tries in a sometimes futile effort to make the thing work. The professional utilizes his knowledge of the laws of nature so that they work to his advantage and not to his defeat. Thus he has a high degree of confidence in the outcome.

And now, gentle reader, having come this far, let me suggest that you stay with me through the remainder of this article, even though some of the material which will be presented seems to be old hat. Just remember that we old timers can say "Oh, I know that!" But there are young lads in the world who will take our place some day and who may not know what we do. It is necessary, therefore, to impart our knowledge to them so that mankind will advance every generation. And besides, even old timers may just happen to learn something! I do, and quite often, too.

THE NATURE OF SOUND

The dictionary is always a good place to start, and Webster defines sound as: "Mechanical radiant energy that is transmitted by longitudinal pressure waves in air or other material medium and is the objective cause of hearing." That seems to say it pretty well.

We all know that the sensation of hearing is produced by pressure waves in the air striking our ear drums and causing them to vibrate. These vibrations are transmitted to the inner ear and thence to the brain where the signals from the ear are interpreted as sound. Thus the mechanism of transmission is by pressure waves. The medium of propagation is usually air (for most all audio work) but it should be noted that sound travels very well through other media such as water, steel, concrete, wood, etc.

The type of wave motion deserves some comment. Physicists recognize three types of waves: *transverse*, *torsional*, and *longitudinal*. A transverse wave is one whose direction of oscillation is perpendicular to the direction of travel. Water waves in a still pond are an example. Electromagnetic waves, such as light and radio signals are of this type. A torsional wave is one characterized by twisting as the wave travels. If one has a very long spring hanging vertically and twists the lower end back and forth in a rotary fashion, waves are set up which will travel up the spring. Sound waves are of the longitudinal type which means that the oscillation is in the same plane as the direction of travel. If we again consider our long vertical spring (which should have come to rest by this time) and pull the lower end down and then release it, a longitudinal wave will travel up the spring to the end where it will be reflected and then travel back down again, etc.

We are going to consider the acoustical properties of rooms, and maybe dispel a few old wives' tales to boot.

VELOCITY, FREQUENCY AND WAVELENGTH

Since sound is a wave motion, it must, therefore, have the same properties as other waves: frequency, wavelength, and velocity of propagation. The frequency of a sound wave is often called *pitch* by musicians, but we will stick to the term frequency.

The frequency, wavelength, and velocity of propagation are related in exactly the same way as for radio waves:

C = (frequency X wavelength)

It should be noted that acoustical writers use the letter C to represent the speed of sound. This speed is, of course, very much slower than electromagnetic waves. It varies with the medium of propagation and the temperature. Since the usual medium is air we should know that at 0°C (+32°F) the speed is 1088 feet per second and at 20°C (+68°F) the speed is 1129 feet per second. At ordinary temperatures the speed can be considered to be 1130 feet per second. which is the value we will use. As a matter of interest, the Smithsonian Tables give the following speeds for other media:

Medium	*Speed (ft/sec)*
Water, air-free, 19°C	4938
Bricks	11980
Iron	16820
Oak wood	12620
Marble	12500

It is evident that sound travels much slower in air than other media and that these media are much more efficient carriers of sound than air. One of the important areas of acoustical engineering is called noise reduction or noise control. This deals with controlling the unwanted transmission of sound, especially through buildings. Those who live in apartment buildings, especially those with cost-cutting construction can testify as to the annoyance produced by sounds from other apartments such as radio and t.v., water running, toilet flushing, and even domestic spats and arguments. Most people who are so annoyed by such noises and sounds do not realize that sound transmission is a two-way street, and their own radio and t.v., toilet noises, and domestic spats also travel to other apartments. Such sounds are transmitted throughout buildings in two ways: by air (air borne), and through the building structure (structure borne). In the usual case, both methods operate simultaneously and must be controlled. But this is another story and beyond the scope of this article. Those interested will find ample material in the existing acoustics books.

Knowing the speed and the frequency, we can determine the wavelength of sound waves. At 1130 Hz. the wavelength is exactly one foot. The wavelengths for other frequencies can be estimated from the values in the table:

Frequency, Hz.	*Wavelength, feet*
30	37.7
50	22.6
100	11.3
500	2.26
1000	1.13
2500	0.453 (5.44 in.)
5000	0.226 (2.71 in.)
10000	0.113 (1.35 in.)

HOW IS SOUND MEASURED?

In order for a scientist to understand a phenomenon, he must know not only *what* is happening, but also *how much* or to what extent something is happening. It is therefore important that one understands how sound is measured so that we can better understand it, and perhaps (hopefully) measure it ourselves for some useful purpose.

Scientists always measure things by their properties, and in sound the two properties most familiar to all are: *pitch* and *loudness*. We have already mentioned pitch as the same as frequency. In acoustical work the term *frequency* is somewhat beclouded by nature and Murphy's Law. In the laboratory, the engineer usually deals with a pure sinusoidal signal and such a signal (if sufficiently pure) will have but one frequency. In acoustical work, however, and especially in noise control, the usual sound source has several or indeed many frequency components. In any case, the idea of pitch or frequency is useful. The frequency of a sound wave is usually measured by picking up the sound with a microphone and then operating on the amplified electrical signal produced by the microphone. We can run it through a wave analyzer, or through a series of electrical band-pass filters, or if reasonably pure develop Lissajou's patterns with an oscillator and cathode-ray oscilloscope (note that 'scopes with triggered sweeps do not lend themselves to Mr. Lissa and Mr. Jou's patterns), or perhaps measure the predominant frequency with an electronic counter. If we cannot haul the laboratory to the sound source (usual case according to Murphy's Law) then we can make a tape recording of the noise and then when we get back, analyze the reproduced tape output.

The *loudness* of a sound is really a subjective term, for as we shall see it depends upon several factors. Yet the term is perfectly valid, for we all know that some sounds are louder than others. But it is still important that we have a measure of loudness so much scientific work has been done on loudness measurement. In general the loudness of a sound depends upon the *sound pressure* developed. Since sound is a longitudinal wave motion, it develops a ripple component of pressure and rarefaction in the ambient atmospheric pressure, in somewhat the same manner that electrical audio signals are a ripple component superimposed upon the direct current passing through an electron tube or transistor. One can visualize this by considering the motion of a loudspeaker cone in creating in front of the cone alternate compression (very slight) and rarefaction as the cone moves back and forth.

The extent of variations in atmospheric pressure is measured in terms of a unit called the *microbar*. Scientists use a unit called the *bar* to denote the pressure of an "average" atmosphere (remember that atmospheric pressure varies with the weather). One bar is 14.504 pounds per square inch or 1,000,000 dynes per square centimeter. Here, the *dyne* is the unit of interest and you will recall that one dyne is the force which produces an acceleration of 1 cm.-per-second per-second when applied to a mass of 1 gram. You will also recall that pressure is always defined in terms of a force-per-unit-area. Since one bar of pressure is 1,000,000 dynes per square centimeter, one microbar is one dyne per square centimeter (just like 1 microfarad is 1/1,000,000 of one farad).

These units are fine, but it is also nice to know what sound pressures are when compared to some standard level, such as the threshold of human hearing. It has been found over the years that a person with "normal" hearing can just hear a 1000 Hz tone whose sound pressure is slightly

In order for a scientist to understand a phenomenon, he must know not only what is happening, but also how much or to what extent . . .

The loudness of a sound is really . . . subjective . . . for it depends on several factors.

greater than 0.0002 microbar (also written as 2×10^{-4} microbar). This pressure, then, has become the internationally accepted reference sound pressure. Since a person also encounters very loud sounds, such as thunder, cannon fire, or jet airplanes, the dynamic range of human hearing is best expressed on a logarithmic basis, and here we encounter our old friend from audio, the *decibel*. The use of dB in acoustic work presents nothing new to an audio man, except perhaps in the definition.

Since we are dealing with sound *pressures* which are usually proportional to the square root of sound power, the sound pressure dB are just the same as "voltage" dB in electrical work. Thus *sound-pressure level* is:

$$\text{dB} = 20 \log_{10}\left(\frac{\text{sound pressure}}{0.0002 \text{ microbar}}\right)$$

Since 80 dB represents a pressure ratio of 10^4 (same as electrical voltage ratio) a sound pressure level of 80 dB represents a pressure of 2 microbars ($2 \times 10^{-4} \times 10^4$) and thus +74 dB (6-dB less) represents a sound-pressure level of 1 microbar. Since 1 bar is 1,000,000 microbars, a sound-pressure level of +74/+120 (ratio of 10^6) = +194 dB would represent a sound-pressure level of 1 bar or 1 atmosphere. Of course no one expects to hear a sound that loud close by, because there would not only be pain and ear damage but also during each cycle the instantaneous pressure would go from a vaccum to two atmospheres!

The measurement of sound pressure in terms of microbars uses a system called the *C, G, S* which means that the fundamental units are the centimeter, gram, and second. There is another system which one finds in acoustic work called the *M, K, S* system which used the meter, kilogram, and second. In the MKS system, the unit for sound pressure measurement is the Newton per square meter and the reference pressure is 2×10^{-5} Newton per square meter (20 micronewtons per square meter). A sound pressure level of +74 dB would be then, either 1 microbar or 0.1 newton per square meter. A sound pressure level of +94 dB would be 10 microbars or 1.0 newton per square meter. Some acoustics texts use the CGS system; others the MKS. We present both to eliminate (?) confusion.

It will be noted that we have begun using the term *sound-pressure level* which is really an absolute measure since it refers to a specific pressure. The situation is the same in electrical work, where plain old dB refers only to a power ratio while the term *dBm* is an absolute amount of power with respect to 1 milliwatt.

There are two other concepts in the fundamentals of acoustics which should be mentioned as they sometimes appear in papers. These are: *acoustic power* and *sound intensity*.

Since sound is a wave motion which requires power for its generation and maintenance, the power in a sound signal may, naturally, be epxressed in watts. Most sounds, however, are of such low power that the watt in acoustics, is something like the farad, too large a unit. Sound power, like electrical power, may be expressed in decibels, either as a ratio or an absolute power when referred to a standard power. The usual designation for acoustic power is *PWL*. Thus:

$$\text{PWL} = 10 \log_{10} W/W_{ref} \text{ (dB)}$$

The literature has apparently not standardized on a reference power, for Beranek uses 10^{-13} watt while the *Genreal Radio Handbook of Noise Measurement* uses 10^{-12} watt. This is not a serious problem as long as the reference power is stated; without this essential information there is a 10 dB error. Using Beranek's reference, it can be seen that an acoustic power (PWL) of one watt is +130 dB.

There is another concept in acoustics which has no analogy in electrical circuits. This is *sound intensity*. Sound intensity is really a measure of energy density, since the unit is watts per unit area. It may be thought of as the rate at which sound energy passes through a unit area in a direction perpendicular to that area. The usual units are watts per square meter or watts per square centimeter. Sound intensity may be put on an absolute basis and expressed in dB by selecting a reference power density. The usual value is 10^{-16} watt per square centimeter which is the same as 10^{-12} watt per square meter.

It must be thoroughly understood that sound-pressure level in dB is *not* the same as PWL in dB, and *not* the same as sound intensity in dB.

We might also mention that there are *no* instruments for directly measuring sound *power* (PWL) or sound *intensity*. Instruments are made and sold for measuring sound-pressure level (ref 0.0002 microbar). These are called *sound-level meters* and consist of a pressure-sensitive microphone, amplifiers, weighting networks which control the amplification at various frequencies, and a meter for reading the sound-pressure level. The values of PWL and sound intensity can be calculated from the sound-pressure meter readings.

This just about wraps up the fundamentals of acoustics from the physics standpoint. In the next gripping episode, we're going to take up how sound is propagated outdoors (inverse square law) (handy for outdoor sound systems) and indoors (that's a different matter) and talk about sound reflection, reverberation, etc. ■

ELLWOOD W. LIPPINCOTT

A Transistorized Switcher for Stereo

For the station that must employ inexperienced combo men that seem to always push the wrong buttons, here is a buildable board to solve these problems.

VISIT any broadcast station and you'll find that today nearly all sources of audio on the air are from two phono turntables, one reel-to-reel tape transport, one or two cartridge machines, one microphone for the announcer, one news-room microphone, and a network line. Other input sources are used from time to time and usually come into the board from a patch panel. In addition there may be automation equipment for all-night operation or for segments of the daily broadcast schedule. All these program sources are controlled by the announcer on duty at the time.

The announcer's main purpose in life, while working, should be to put the best sounding, cleanest signal on the air with the least number of "goofs". His on-the-air duties should not include production or auditioning. He must re-create the sounds already produced on record, tape, cartridge, or from the news room or network. He must also announce each program event as it happens. In order to perform his duties he must have the equipment that will cause the least frustration to both him and the listener.

Many broadcast stations, particularly those in the smaller communities, use inexperienced help as announcers. Frequently this is a part-time employee who is attending a local high school or college. This is not always the best plan, but let's face it, it is economical.

What can management and engineering (they must work hand-in-hand for a successful operation) do to make this announcer's work on the air more professional–make it sound as the large metropolitan stations sound.

Frequently the announcer or the engineer will suggest a new audio console–one with sixteen mixer controls, twenty or thirty inputs, several gain controls, many lights and switches. The suggested console might even include equalizers, compressors and echo. In the end it will probably look like the control panel of an SST.

As stated earlier the announcer's job is to re-create what has already been produced. It is my opinion that mixers are not needed. Two or more sources of audio at the same time are confusing. If the musical background is interesting the commercial message looses its drive. If the message is interesting then the music is lost. Most listeners can not or will not listen to two things at the same time with any degree of interest so the musical background becomes incidental and of no value except to detract from the complete thought of the sponsor's message. If for some reason it is necessary to broadcast music, and talk at the same time, it should be recorded by the production department. The announcer on duty will then play the tape or cartridge. Remember, the announcer should introduce a program event, start the event, adjust gain if and when necessary, wait until the event comes to an end, announce and then start the next program event. This should constitute the announcer's duties throughout his time on the air.

It is my belief that the announcer's work should be made as easy as possible, not because I feel he is overworked but because the less he has to do the fewer mistakes he will make. He should have time to organize his thoughts, study the commercial or other announcements he is to make live, and prepare himself to present a smooth flowing, listenable program.

With this thought in mind and with over 44 years experience as an announcer/engineer I have designed what I feel is the ideal audio system for a station using combination men as announcers. This is particularly true with a station broadcasting classical music. This system or board consists of ten lighted and labelled pushbutton switches for switching ten program sources and one slider type double L attenuator for gain control. There are also four lighted and labelled pushbutton switches for stereo or mono mode and for live or automated operation. Such a board is shown in *Figure 1*. A simplified single line drawing of the entire system is shown in *Figure 2*. All components are labelled the same as in the over-all system drawing, *Figure 3*.

It is my belief that the fewer the components the less the chance of equipment failure. Also, the lesser number of component types makes for easier maintenance. All possible components should be plug-in for easier maintenance. The audio switcher to be described uses only one type of relay (Potter & Brumfield type KPH) and one type of amplifier. The amplifier is a Melcor 1731 (or similar) and all equalizing and other small components are mounted on the amplifier socket. Technical information will be gladly supplied by the manufacturer. Pick-up and tape-head load resistors are 50k ohm audio taper potentiometers and all input resistors for gain adjustment are 600-ohm potentiometers.

Momentary contact switches, S2 through S8 of *Figure 3*, should be designed so that the contacts used to operate relays K1, K2, K3 and K4 close after other contacts of these switches have closed. In this way, preamplifier-input switching and motor-starting relays are closed before the switching of the preamplifier outputs to the audio busses by relays K1, K2, K3 and K4. An alternative would be to select the four slowest operating relays for K1 through K4.

Ellwood W. Lippincott has spent over two decades in front of and behind combination boards.

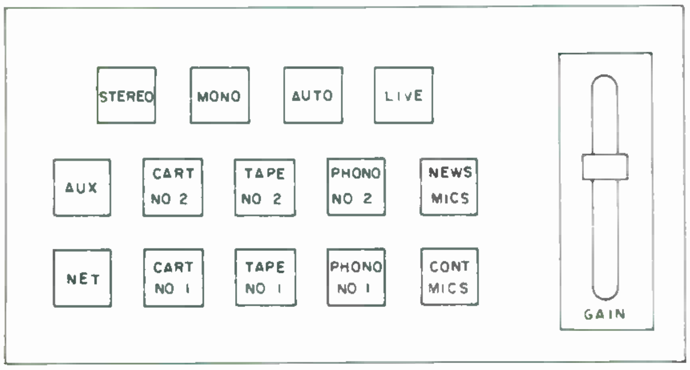


Figure 1. The author's design of a combination-man audio board.

Two power supplies are required. One dual unit supplying 15 volts positive and 15 volts negative for the integrated-circuit amplifiers. This supply should be capable of 250 to 300 miliamperes minimum per section. The second power supply is only used to operate the relays and the lights in the switches. Although the drawing calls for 110 volt d.c. relays and neon lights. any desirable units may be used. All relays are shown in the non-operating condition.

As shown, there are two control-room and two news-room microphones. Either the news room or control room may be used for interview programing. There are also facilities for two stereo turntables. two reel-to-reel stereo tape transports, two mono cartridge players. one network line. and one auxiliary input which can be expanded to any number desired (See *Figure 4*). Provision for either stereo or mono automation programing has also been provided. If mono only operation is desired, the circuit can easily be modified.

The gain from each program source is screwdriver adjusted behind the board by the engineer so that the level from each source is of equal value in normal operation. When a program event ends the announcer pushes the momentary contact switch labeled *control microphone*. This will connect the two control-room microphones (right and left) to the outgoing audio busses with the gain already adjusted to the proper level, and will disconnect all other audio sources while the microphones are on. If the level of the announcer's voice is too high or low a slight correction is made by adjustment of the double L attenuator *gain* control.

When the announcer completes what he has to say and introduces the next program event he starts that event by pushing the proper switch button. This will automatically disconnect his microphones from the audio busses and connect the source for the event desired as well as start the phono turntable. tape transport, or cartridge machine being used.

In stereo broadcasting it is desirable to use both a right and a left microphone for a single voice. This has been provided for in both the control room and the news room. I prefer favoring either the right or the left microphone when only one voice is used. In the system being described the left microphone should be favored. The reason is that the cartridge machines (mono) connect to the right channel only. With this type of operation all live announcements seem to appear on the left channel while all taped (cartridge) announcements appear on the right channel. The listener is more aware of the stereo effect when program material (single speaking voice) alternates from right to left channels. If favoring the right or left channel is to be practiced, it should be accomplished by microphone placement rather than the adjustment of channel gain. When setting up the microphone levels, both microphones should be placed side by side and the screwdriver-adjusted potentiometers R1 and R2 set so the gain is equal in both channels. When a two-microphone interview is being broadcast the gain of the right and left channels should be approximately equal by microphone placement.

Since networks feed news programs on a mono basis and since local remote programing (live) is also supplied on a mono basis both the *network* switch button and the *auxiliary* switch button automatically connect the system for mono operation. That is, the right and left channels are combined and fed to the transmitter on the left channel only while the stereo generator is made inoperative. This is accomplished by contacts on relays K5 and K7, or by K6 and K7 of *Figure 3*.

Before going into a detailed description of the over-all circuit and operation of the audio switcher it might be advisable to point out some of the inovations included and the reasons for including them.

Cuing has been deliberately avoided—in fact it has been made impossible. The reason? First of all, records can be set with the stylus on the outside grove, revolved one or two revolutions and be ready to play flawlessly. As stated earlier, the announcer does not or should not audition while on duty. I have visited a classical music station and found the announcer listening to a Bill Cosby record on the cue amplifier while broadcasting a Brahms symphony on the air. At least he *thought* the symphony was being broadcast.

Both phono turntables and pick-ups feed the same preamplifiers making cuing of records impossible. Cuing of reel-to-reel tapes can be accomplished with head phones when setting up taped programs for broadcast. Cartridge machines are self cuing. Networks feed on time so a correct clock is all that is necessary. The same can be accomplished with live remotes from the church or football field. I do not feel cuing is at all necessary and in many cases it is undesirable. Audio monitoring is accomplished by feeding the "on air" signals from the modulation monitor to the speakers. VU meters should be connected across the outputs (right and left) of the switcher.

Something unusual in audio systems has been incorporated in the method of amplifying phono, tape, and cartridge pick-ups. Two preamplifiers (right and left channels) have been provided for the phono pick-ups with the inputs of the two amplifiers switched to either the number 1 or number 2 turntable. Common practice would call for four rather than two preamplifiers. The same arrangement, with two preamplifiers, has been provided for the reel-to-reel tape transports, the newsroom and the control-room microphones. and one preamplifier for the two cartridge machines.

Referring to *Figure 3*, two microphones are connected to the points labeled *cont mics L & R* on relay K9. and two microphones are connected to the points labeled *news mics L & R* on relay K9. The outputs of the left and right phono preamplifiers should be connected to the potentiometers R3 and R4. The reel-to-reel tape preamplifier outputs connect to R5 and R6. The preamplifier output for the two cartridge machines connect to R7. The incoming network line connects to R8 and the auxiliary input, or a system such as shown in *Figure 4*, connects to R9.

By operating momentary-contact switch S1, power is applied to relay K1 connecting the outputs of the microphone preamplifiers, IC1 & IC2, through potentiometers R1 & R2 to the inputs of line amplifiers IC3 and IC4. The outputs of the line amplifiers are now connected through the n.c. contacts of K8 to the L and R outputs of the switcher through the 600 ohm-to-600 ohm ladder attenuators R10 and R11 used for over-all gain. Power from point W on K1 is also applied through the n.c. contacts of K9 to the switch light LS1 and to K15. Through the operation of K15 the control room speakers are disconnected from the monitor amplifiers and load resistors R12 & R14 are connected. Other contacts of K15 are

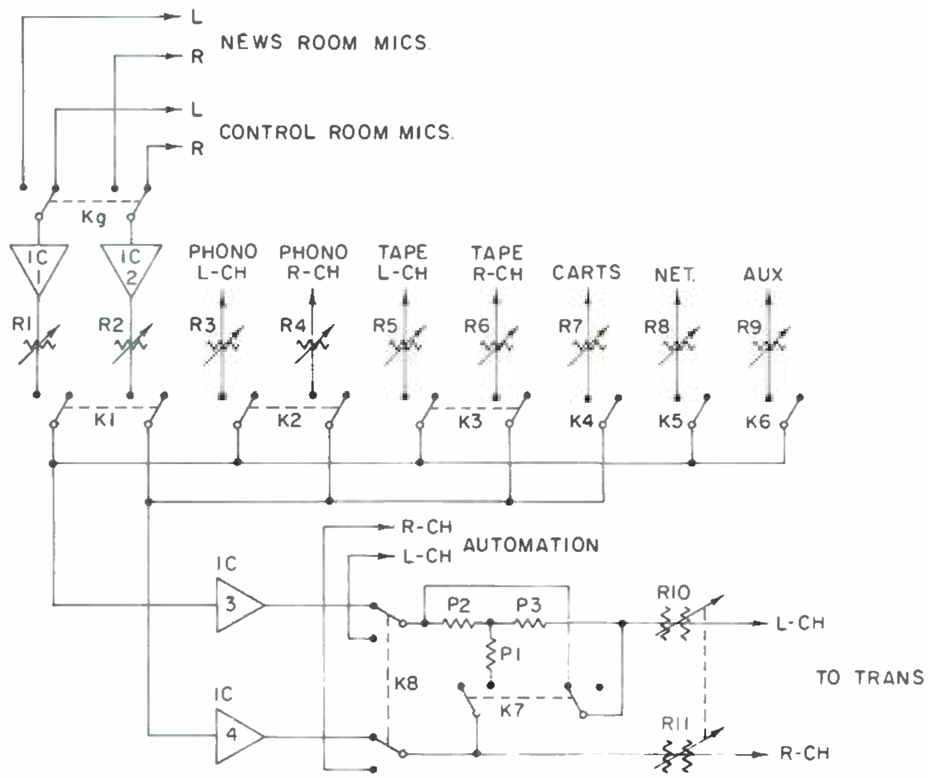


Figure 2. A simplified single line drawing of the board shown in Figure 1.

parallel-connected and operate an on-the-air light.

The L and R microphones in the control room are now connected through the preamplifiers IC1 and IC2 and the level-adjusting resistors R1 and R2 to the inputs of line amplifiers, IC3 and IC4. The outputs of the line amplifiers are now connected, through over-all gain controls R10 and R11 to the transmitter lines or to the gain-adjusting amplifiers and the announcer can talk on the air. A lighted switch button, S1, reads *cont mics*, the control speakers are muted and an *on the air* light is lighted outside the control room. Through the two lower sets of contacts on K1, all program sources except the control room microphones have been disconnected and locked out of the inputs to the line amplifiers.

To put the news room on the air, the announcer only has to operate momentary contact switch S2 and the exact condition exists as with S1 except the second section of S2 applies power to relay K9 connecting the news-room microphones to the microphone preamplifiers instead of the control room microphones. Lamp LS2 now lights switch S2 while S1 is darkened and K16 is operated instead of K15. By this operation the news-room speaker is disconnected from the monitor amplifier, a resistor is substituted and *on the air* lights are lighted both inside and outside the news room.

Operating momentary contact switch S3 applies power to relay K2 and through the second section to K13. At the same time, through contacts on K2, power is disconnected from the operate coils of all other relays, and the left and right outputs of the phono preamplifiers are connected through the level adjusting potentiometers R3 and R4 to the left and right line amplifiers, and through them to the output of the switcher. Contacts on K13 also apply 110-volts a.c. to the number 1 turntable setting it in motion. S4 starts phono motor number 2, switches the input to the phono preamplifiers through K10 and lights the S4 switch button.

Momentary contact switch S5 operates in the same manner as S3 with the exception that the additional contacts on S5 will start reel-to-reel tape transport number 1. S6 starts tape transport number 2 and switches the pickup heads of the transport to the preamplifier inputs.

Through the same type of operation, S7 sets cartridge transport number 1 in operation and connects its output to the right channel *only*. S8 performs the same functions for cartridge transport number 2.

S9 connects the network through relay K5 and at the same time operates K7. Through the operation of K7 the entire audio system is switched to the mono mode by combining the outputs of the left and right channels in the resistor network (P1, P2 & P3) and through contacts on K7 the stereo generator is made inoperative.

Switch S10 performs the same functions as S9 except in this case it is for the auxiliary input or inputs.

Since relay K7 is usually in the stereo (non-operate) condition the audio system can be put in a mono mode by S12 and returned to stereo mode by switch S11, a normally-closed momentary contact switch.

Switches S13 and S14 connect the proper source, local switcher or automation equipment, through the double ladder attenuators, R10 and R11, to the gain-adjusting amplifiers or to the transmitter lines.

The load resistors for the phono pickups, 50k-ohm audio taper with screwdriver adjustment, potentiometers R, R1, R2 and R3 of *Figure 5*, are adjusted as near maximum as possible and with equal output of each pickup and channel as measured across gain-adjusting resistors R3 and R4 of *Figure 3*.

The dual-ladder gain controls are set at approximately mid scale and input (gain-adjusting) potentiometers, R3 and R4, are adjusted so that the level of each channel is equal and of sufficient gain to drive the gain-adjusting amplifiers or the transmitter lines. R10 and R11 remain at this mid-scale setting.

The same process is used in setting the load resistors, R, R1, R2 and R3, and the input resistors, R5 and R6, for the reel-to-reel tape transports. With tape, the head-loading resistor should be 300k ohms. By using a 50k-ohm audio taper potentiometer in series with a 250k-ohm fixed resistor the proper load resistor is achieved.

Tape-head load resistors and the input resistor, R7, for the tape cartridge machines are adjusted in the same manner as for the reel-to-reel machines.

Network level is adjusted by R8 and the auxiliary input (if only one auxiliary input is used) is adjusted by R9. If more than one auxiliary input is provided, using the circuit

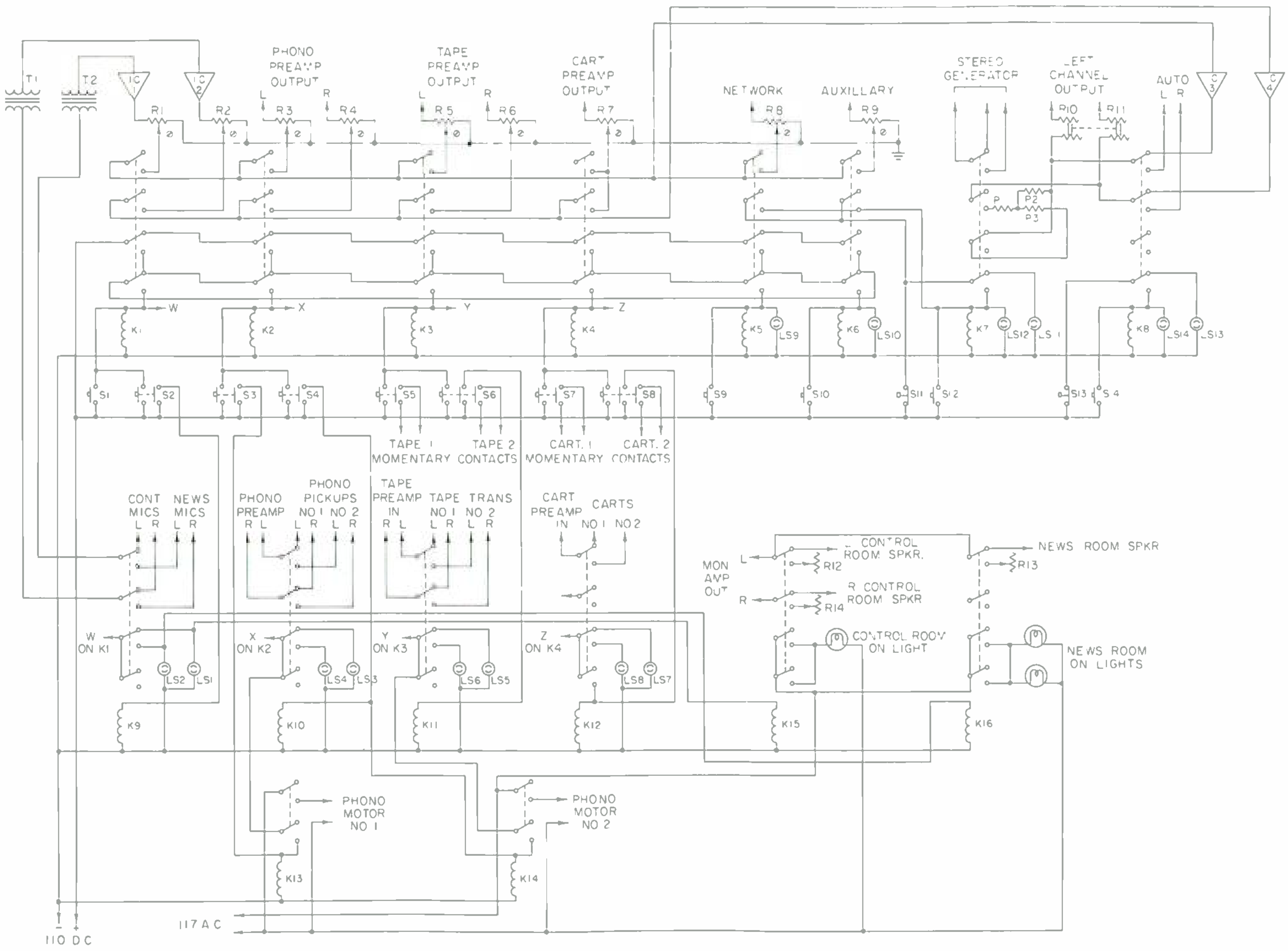


Figure 3. The switcher board in detailed schematic.

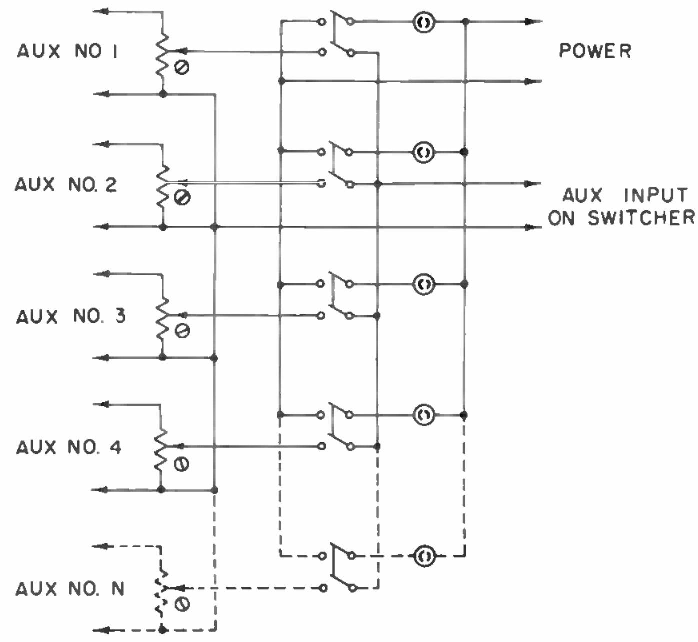


Figure 4. Additional switching facilities for the board.

of *Figure 4*, the input resistor R9 should be disconnected as each auxiliary input has its own level setting potentiometer.

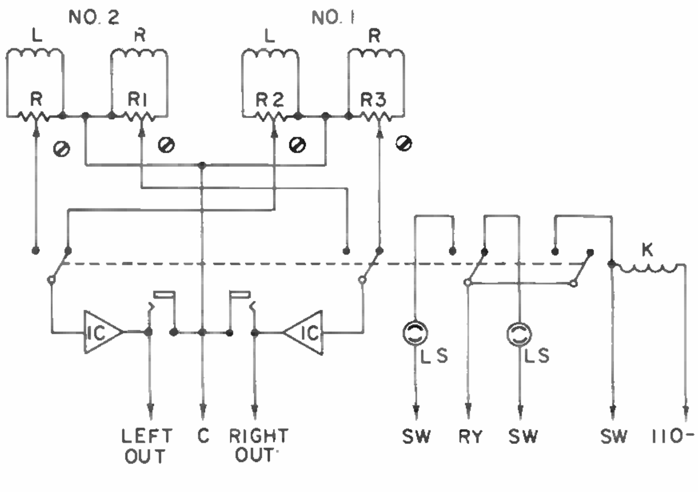
To set the microphone levels place both microphones in the control room as close together as possible. When speaking in a normal voice adjust input resistors R1 and R2 so that the gain of each channel is equal and of the proper level at the output of the switcher. Once adjusted, the right-channel microphone should be moved so that the majority of pickup is with the left channel. Any change of level between various announcing voices should be corrected by moving closer or farther from the announcer.

The levels of various voices, records, tapes, cartridges, etc. are not always exactly the same. The increase or decrease in gain necessary for smooth operation should be taken care of by a slight adjustment of the dual ladder gain controls R10 and R11.

The audio system and switcher described should prove to be a real boon to the small community broadcast station that is forced to use inexperienced help. The maintenance engineer will find servicing is extremely easy since all amplifiers and relays are plug in. Replacing of a defective unit requires only a matter of seconds.

An inexperienced announcer can learn the board almost instantly and can effortlessly acquire metropolitan production of programs with an excellent frequency response and a low distortion. He will also find it almost impossible to make operating errors. ■

Figure 5. The phono pickup load resistors are adjusted as near maximum as possible and with equal output of each pickup and channel.



MARSHALL KING

A Day in t.v. Audio

Sometimes when we think a utopian recording condition has arrived, Murphy's law takes over to prove itself the real master.

I show you the picture of *Figure 1* just so you will have the extreme pleasure, as I did, of seeing the ultimate in perfect conditions for recording sound. That endless dry lake you see there in the middle of the Mojave Desert is about as perfect a chamber (anechoic) as a guy could want: miles and miles of nothing but cracked and hardened mud which has been left behind by time.

. . . it was just as I thought: absolute silence.

Figure 1. The author looks over the completely barren El Mirage Dry Lake at the beginning of a day of t.v. recording.

. . . I was all set to enjoy, for the first time in my life, test-tube conditions in audio.

Sitting atop a step-ladder at six in the morning, waiting for the rest of the cast and crew to show up so we could get on with the business of shooting a television commercial on the new Ford cars for the then-coming season, I scanned the horizon in all directions to see if somewhere in the distance there might be a groundhog or a butterfly flitting about in a noisy manner. But no, it was just as I thought: absolute silence. The only threat was that the immense blue sky was punctuated by a small fluffy cloud about eight miles away which possibly could make some sort of cloud sound as it went by. I made a mental note to keep my eye on it. Other than that, I was all set to enjoy, for the first time in my life, test-tube conditions in audio.

What a thrill! What sound man, burdened with the unmitigated racket of air conditioners, noisy cameras, and the squeaky shoes of impolite ushers, all bouncing off the confining floors and walls of a television studio, hasn't dreamed of such ideal conditions? You must know that I considered myself a lucky man when this day began.

But that was at six in the morning. Before the day was over I was a wreck, utterly thankful to have gotten something, *anything*, on tape. And I got all kinds of things, along with some of the sounds I wanted. In short, I re-learned what I've always known: there will never be perfect conditions for recording sound in television. Although we always come out with a fairly decent, if not spectacularly good, product, the truth is that all remotes are handled as though television was invented just this morning and we're going to try it out this afternoon. (You tell me why this is so). My test-tube conditions went astray, and it wasn't because of a fluffy little cloud.

The commercial was to be handled as follows. Our announcer, Jim Makrell, is shown in a medium close-up standing in front of a huge circle of new Fords, as the desert in the background goes off into infinity in all directions. During the nine seconds it takes him to say his first two sentences, the camera, which is mounted on an 85-foot crane, starts at eye level and moves up, up and away . . . so that by the time Jim has said four sentences the camera has gone high over his head and beyond him, and remains looking down at the circle of cars during the rest of the message.

Marshall King is audio supervisor and sound mixer at the Hollywood Video Center.

Figure 2. The new Fords and the t.v. equipment are rolled into place while the day is still young.

Figure 3. An attempt was made to record Jim Mackrell's voice while the generators were running, but the effort was abandoned.

Now pay attention folks, while I tell you the requirements of the director. He wants to do this in one continuous shot, without any stops or cutting. He does not want to see a microphone or a cable in the picture. Since we are starting with a close-up on the announcer, we cannot resort to *pre-record* and *lip-sync.* He will allow a wireless mic to be used, provided it is of recording-studio quality. And he wants to begin shooting when the sun proscribes a subtended angle of thirty-seven degrees with the vernal zenith. Which means he'd like to begin shooting right now. You may as well know that we did *not* begin shooting right now. We didn't begin shooting for another five hours due to a variety of reasons, but we'll concern ourselves here with the audio problems.

The biggest reason, of course, was that due to the close-up shot of the announcer we could not use a pre-record. For after all, no actor in the world can accurately lip-sync to a spoken dialogue track for more than a word or two. Music and rhythm, yes. Unmetered dialogue, never. Now if the director had not insisted on doing the thing on one continuous shot, I could have put a microphone just off camera during the close-up, then stopped the camera while we got rid of the mic and hooked up our playback facilities, then continue with the long shot using pre-record and lip-sync. Slipshod lip-sync at that greater distance wouldn't be detected anyway. But no, we had to start with a close-up and stay on that camera with no stopping as the camera made its climb for the aerial shot.

Since we are starting with a close-up on the announcer, we cannot resort to pre-record and lip-sync.

Figure 4. Coordinator Don Repke, actor Jim Mackrell, and author Marshall King pre-record the audio at a spot far removed from the generator noise.

. . . it's not up to the audio man, or anyone else in the technical crew, to suggest . . . how to re-write their scripts to accomodate physical problems and equipment limitations.

Now it's not up to the audio man, or anyone else in the technical crew, to suggest to the production people how to re-write their scripts to accommodate physical problems and equipment limitations. In fact, Cecil B. DeMille seemed to take a particular pleasure in not allowing his people to tell him of the vast number of impossibilities he had conceived. Of course, he always had the last word by turning out one epic after another. This is all very well for the movies of three decades ago, but today in television things are geared to work right the first time (they never do), and to be put on the air tomorrow (they always are). There are no rushes or dailies in television; there are no midnight conferences to correct the day's snafus; there are usually no machine shops or carpenter compounds where weeks are allowed to create Mary Poppins magic. If it can't be rented, transported, rigged, and operated in half a day we can't use it. Furthermore, the nature of television production, while not yet through emerging into an identity that may be definable, is now a concept whereby the keyest of key people are obtained (hopefully) to form a crew for a day, so that most t.v. units outside the networks consist of ever-changing groups of highly experienced strangers who are forever on the move and trying to remember where, if at all, they have seen one another before. It's as though the people who put these things together feel that by hiring the most knowledgeable crew they can find, they have precluded the slightest possibility of an impossibility. Such is not the case, as this day nearly proved.

Figure 5. The proper solution was to record the audio in the bus shown at the left, after it was moved away from the generator truck seen at center. The t.v. truck just poking into the picture at the right was used as a control room where both picture and sound were laid onto video tape.

All during those first early morning hours while the camera and lighting people were setting up their equipment, an inner voice kept telling me that we should be using this "quiet" time to pre-record the announcer, just in case. For, if they changed their minds and decided to lip-sync after all, it would be too late to pre-record once the generators were running. Those of you accustomed to motion-picture techniques will see this as much ado about nothing, for all one has to do in films is to loop the sound back at the studio at a later time. But on the other hand, film work cannot possibly meet the time requirements which television does. This commercial (like most all others done in t.v.) had to be viewed after each take, had to be finished by the time the sun went down, edited by tomorrow night, approved by the client within two days, and on the air this coming weekend. Beyond all this, Jim Mackrell, whose voice they wanted, was leaving for another assignment in the morning. *What* looping?

You have a right to ask: didn't someone think of all this before the day began? The answer to that is: someone thought of it and came up with the old magic words that directors love so much . . . . *Wireless Microphone.* I have said about all I want to say regarding wireless microphones in an earlier article. In brief, they are improving all the time, which is rather phenomenal considering that the FCC has given broadcasters neither sufficient wattage nor the exclusive practical frequencies needed for studio wireless microphone work. I was incredulous to find that a man's breathing as he set foot on the moon could be heard clearly 235,000 miles away, while I get wiped out by every passing cab or police car when I operate a wireless mic more than twenty feet from its receiver.

Nevertheless, I was prepared to use a new Vega on actor Jim Mackrell, for quite frankly there was no other way to do it. I wasn't worried about any passing cabs or police cars out here, but I was worried that *any* kind of mic would pick up the sound of our heavy-duty gasoline-driven generators which supplied power to everything we owned . . . . cameras, lights, audio console, and coffee urn. We had enough umbilical cord to put the generator two hundred feet from the microphone, but it may as well have been two feet, for that gigantic, smooth, slightly parabolic clay dish we were sitting on acted as a perfect sound carrier. You know how sound drops dead, out on the wide open spaces, after travelling a few feet? Not on this dry lake.

There was no way to hide the roar of the generator, but we had to make several attempts at recording to prove the point.

Figure 6. The playback speaker used to cue the actor for lip-sync is seen at the end of the cable at the left. Even this position was too close for the camera, and the speaker had to be moved back another 15 feet.

There were no way to hide the roar of the generator, but we had to make several attempts at recording to prove the point.

The point was proved, and producer Lou Jacobi called a halt to everything as he decided that we must pre-record the audio after all and that Mackrell would have to lip-sync the first few lines. So, along with coordinator Don Repke of J. Walter Thompson, Jacobi jumped in a station wagon with me and a battery-operated tape recorder, and of course we took Mackrell, and headed out across the boondocks to find a spot where the generators could no longer be heard. Why not turn the generators off and make the tapes right there on the scene? It would have meant the cameras would have to be aligned all over again when the power was restored, which would have meant a further delay. Now, time was becoming a serious factor, for the sun was over the yardarm and moving steadily.

We tested the acoustics here and there, and came to rest a mile away. Our cameras and trucks and generators were small objects halfway to the horizon, and we were satisfied that quiet prevailed. I used myself as a mic stand and held an ElectroVoice 635 for Jim, and he gave us a flawless reading the first time around. Unfortunately, the last sentence was covered by the drone of a jet. When it passed we tried it again, but this time Mackrell booted a word. Take three, another jet. Edwards Air Force Base, where Major Chuck Yaeger broke the sound barrier some years back was just over the hill (if there had been a hill). Producer Jacobi stopped the readings of the next several takes in midstream, for now there was a discussion of the wording. Takes nine through fourteen ..... more high flyers. Take fifteen, and Mackrell loused up his pacing while listening for another jet which never came. It came on the next one instead. After take nineteen I knew I was in some kind of sinister contest between myself and the U.S. Air Force. I imagined all the secret radar-laser-type equipment which lay in the Edwards control tower, all of it focused on us and our tape recorder, so that every time we got speed the word flashed throughout the air base for another jet to take off. But, in time, I sensed a pattern in their take-offs, as though I had developed ESP right on the spot. After awhile I simply knew when they were going up, even though we seldom saw one.

After take nineteen I knew I was in some kind of sinister contest between myself and the U.S. Air Force.

On the last words of take twenty-four the sound of an approaching car destroyed our efforts. Unit manager Rich Zarro leaned out of his car window and asked how much longer we would be. We whipped off our belts with the intention of thrashing him severely about the head and shoulders, but he made his escape back to the camera area.

By now the desert, which this morning had been so silent, so desolate and quiet, had become a rumble of activity and noise. In addition to the jets from Edwards, there were a variety of sounds from gasoline-driven engines, and somewhere in the distance I thought I heard a baby crying. I could accept that, for at this point I was ready to believe anything.

Suddenly I became aware that El Mirage Dry Lake was being crisscrossed and circled by many lovable youngsters of about thirty, beards flying in the breeze, who zipped hither and yon astride various kinds of motor bikes. Since there seemed to be more appearing on the scene at every moment, I suspected what must be happening. You've heard of love-ins, drop-outs, put-downs, goof-offs, come-ons and lock-ups? Well these flower children were obviously holding their annual Noise-Over. I signalled one to a stop, and asked him if he could give me an estimate of how long we might have the pleasure of his company. He didn't comprehend a word I said, as he sat there breathing heavily through a mouth hanging idly open. He studied my lips as I spoke, as though by so doing he might discover the meaning of the sounds I was making. Poor chap, I couldn't help feeling sorry for him as he stared at me numbly and raced his engine; certainly he was in no way to blame that his genes had slipped through the fingers of evolution. When he was convinced that I wasn't going to give him any raw meat or loose change, he roared away to join his tribe, not even noticing that his autographed picture of Peter Fonda fell out of his back pocket. I just prayed that he didn't get his hair caught in his rear sprocket.

By this time we were in bad shape, for we still didn't

Figure 7. At this height the camera can't tell if the actor's lips are out of sync, but it can see the playback speaker and its cable if care is not taken.

have one acceptable reading, and I was assured that there was no time for me to edit the clean parts together to make one good take. But we got it on take thirty-one, and hurried back to the camera area where almost everyone was sitting around waiting for us. The cherry-picker (the 85-foot camera crane) had been rolled into place and the new Fords were in position. To emphasize that everyone was waiting for audio, and that they were well aware of my troubles, two cameramen were putting on a tableaux for my benefit. When I stepped out of the station wagon one of them was sitting on the shoulders of the other, looking at the vast dry lake through a pair of binoculars.

"Hold still now, I think I see it," he cried. "Yes, there it is! It's either a pink titwillow or a white-breasted dowager flapping its wings in a noisy manner and coming this way. Whatever will the sound man do?"

The mirth didn't last too long, for within moments we received the worst news of all. Someone arrived from our Hollywood studios with new copy from the client, so that all our recording had been for naught. 'Tis a pitiful thing to see a grown man cry, so I turned my head away in an act of courtesy in case one of them broke down. But no one panicked, and Lou Jacobi made a quick and final decision. "That does it," he said. "We'll re-record Jim right now, over there inside that bus, and we're not turning off any generators, either. Let's go."

It's nice to have a nasty decision taken out of your hands by a big man, and though I was apprehensive I felt relieved. We moved the bus a little further away from the generator truck, stuck Jim inside with his new copy, sealed the doors and windows except for a crack which let the mic cable pass through, and we accepted what he read on the first take. We had to: the sun was getting dangerously low and we had not yet put a thing on video tape. And what we had got on audio tape was not particularly thrilling to me, for in my mind's ear I could still hear the clean, open sound of the unattended desert, which we almost got earlier in the day. Yet, Jacobi was right, for the bigger problem had to be considered. Sound was just one part of it; he still had his pictures to get.

The one remaining problem was solved just as quickly. When it was seen that lip-syncing dialogue was really too much to expect of any actor, they changed Mackrell's position so that the back of his head was seen during the first few words. That gave him a chance to "catch up" to the dialogue as he turned around to face the rising camera just before his head disappeared out of lower frame. It was a compromise that no one had wanted to make, but then, recording the sound inside a bus was also a compromise. The disappearing sun had been our merciless dictator.

I saw the commercial in our studios later at Hollywood Video Center, and, to be honest, coming back to it after a day or two, it looked and sounded quite good. I often wonder, considering the tons of racked up footage that is put on television every week, just how many compromises are made by someone, somewhere, in all cases. It may be true that we appear to carry on as though television was just invented this morning, but considering the quality per mile, I don't think we should be ashamed. We can be proud not to have gone insane, if nothing else.

On this day that I'm talking about, we got what we wanted before the sun went down, and everyone headed back to the city. The lighting guys were the last to strike, and while waiting for one of them to give me a ride home, I climbed up on the ladder they had brought along, and I surveyed the endless desert. All activity had stopped, and with the setting sun a deathly silent loneliness crept in from all directions. I couldn't help thinking what a fine spot this would be for the perfect audio recording, if someone wanted to do a remote here. ■

Circle 29 on Reader Service Card

BOOKCASE

As a service to our readers we are pleased to offer these books from prominent technical publishers. All prices listed are the publishers' net. Shipping charges are included.
Use the coupon at the bottom of the page or give the complete title, author and coupon number. Be sure to indicate quantity on the special instructions line if more than one copy of a title is wanted. Full payment must accompany your order. We cannot ship c.o.d. Checks or money orders should be made payable to Sagamore Publishing Company, Inc. Because of the time required to process orders, allow several weeks for the receipt of books.

Radio Engineering Handbook, 5th Ed.
Henney. Prepared by a staff of specialists, this working manual of radio science provides information on each of the branches of radio engineering, with emphasis on working practice, final working formulas, dimensions, and actual useable circuits.
1959. 1,800 pp. **$31.50**
Circle 36 on Coupon Below

Radio Transmitters
Gray and Graham. Provides, in a logical, easy-to-understand manner, a working knowledge of radio transmitters for quick solution of problems in operation and maintenance.
1961. 462 pp. **$16.00**
Circle 40 on Coupon Below

Electronic and Radio Engineering, 4th Ed.
Terman. A thorough coverage, in easy-to-understand terms, of those principles and techniques which are the basic tools of the electronic and radio engineer. 1955. 1,078 pp.
$19.50
Circle 39 on Coupon Below

The Technique of the Sound Studio — **NEW EDITION**
by Alec Nisbett. This, is a handbook on radio and recording techniques, but the principles described are equally applicable to film and television sound. 264 pages; 60 diagrams; glossary; indexed; 5½ x 8½; clothbound. **$13.50**
Circle 1 on Coupon Below

Radio Broadcasting
edited by Robert L. Hilliard. Each of the five chapters has been written by a prominent educator with an extensive background of practical experience in commercial and educational broadcasting. 190 pages: 6¼ x 9½; indexed; clothbound. **$6.95**
Circle 2 on Coupon Below

Transistors for Audio Frequency (Audio-Frequency Amplification)
by Guy Fontaine. 1967. This systematic and detailed treatment of the application of transistors in audio-frequency amplifiers shows how the published transistor characteristics are related to the principles of design. An ideal textbook or reference on the subject for engineers and advanced technicians. 384 pages; 5½ x 8; illus.; clothbound. **$7.95**
Circle 12 on Coupon Below

Closed-Circuit Television Handbook
by Leon Wortman. Gives comprehensive detailed information about the field in an easy-to-understand presentation, suited to those who plan to use, install, and service cctv. 288 pages; 5½ x 8½; clothbound. **$5.95**
Circle 18 on Coupon Below

An Alphabetical Guide to Motion Picture, Television, and Videotape Production
Levitan. This all-inclusive, authoritative, and profusely illustrated encyclopedia is a practical source of information about techniques of all kinds used for making and processing film and TV presentations. Gives full technical information on materials and equipment, processes and techniques, lighting, color balance, special effects, animation procedures, lenses and filters, high-speed photography, and much more. 1970. 480 pp. **$24.50**
Circle 35 on Coupon Below

Integrated Circuits
Motorola. This complete, authoritative volume of design and construction techniques for modern integrated circuits covers theory, thin-film techniques, diodes, transistors, thermal design, and other vital aspects.
1965. 373 pp. **$13.75**
Circle 30 on Coupon Below

Basic Bibliography of Science and Technology
McGraw-Hill. Important to the researcher, librarian, or student, this reference book lists and describes over 8,000 outstanding books in every scientific and technical field. A topical index organizes all subject headings into about 100 general categories enabling the reader to locate quickly the listings which pertain to his field of interest.
1966. 738 pp. **$19.50**
Circle 34 on Coupon Below

Noise Reduction
Beranek. Designed for the engineer with no special training in acoustics, this practical text on noise control treats the nature of sound and its measurement, fundamentals of noise control, criteria, and case histories. Covers advanced topics in the field.
1960. 752 pp. **$19.50**
Circle 33 on Coupon Below

The Audio Cyclopedia (2nd ed.)
by Dr. Howard M. Tremaine. New and updated, here is the complete audio reference library in a single volume. It provides the most comprehensive information on every aspect of the audio art. This new edition includes the latest audio developments including the most recent solid-state systems and integrated circuits. It covers all subjects in the fields of acoustics, recording, and reproduction with more than 3400 related topics. Each topic can be instantly located by a unique index and reference system. More than 1600 illustrations and schematics help make complicated topics masterpieces of clarity. 1760 pages; 6½ x 9-3/8; hardbound.
$29.95
Circle 20 on Coupon Below

Circuit Design for Audio, AM/FM, and TV
Texas Instruments. Texas Instruments Electronics Series. Discusses the latest advances in electronic design and application which represent the results of several years research and development by TI communications applications engineers. Emphasizes time- and cost-saving procedures throughout. 1967. 352 pp. **$14.50**
Circle 32 on Coupon Below

Acoustics—Room Design and Noise Control
by Michael Rettinger. 1968. The enormous problems and hazards presented by noise are dealt within an orderly and practical manner. With many charts, graphs, and practical examples, the text covers the physics of sound, room acoustics, and design, noise and noise reduction. 392 pages; hardbound. **$17.50**
Circle 21 on Coupon Below

Sagamore Publishing Company, Inc.
980 Old Country Road, Plainview, N. Y. 11803
Please send me the books I have circled below. My full remittance in the amount of $.......... is enclosed. N.Y. State residents add 5% sales tax.
1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16 17
18 19 20 21 22 23 24 25 26 27 28 29 30 31 32
33 34 35 36 37 38 39 40

Name ....

Street Address ....

City .... State .... Zip.....

Special Instructions ....

Canada and foreign: Add $1.00 per book

Acoustical Tests and Measurements
by Don Davis. Provide; a solid understanding of the entire subject of acoustical measurements; based on actual field test work, using commercial equipment. 192 pages; 5½ x 8½; hardbound. **$6.95**
Circle 7 on Coupon Below

Television Broadcasting
Chinn. A practical, compact technical guide to the equipment, systems, facilities, good engineering practices, and operating techniques of television broadcasting. 1953. 695 pp. **$17.00**
Circle 37 on Coupon Below

Handbook of Electronic Tables & Formulas, (3rd edition)
A one-stop source for all charts, tables, formulas, laws, symbols, and standards used in electronics. Includes an 8-page, full-color fold-out chart showing latest FCC allocations for the entire frequency spectrum. 232 pages; 5½ x 8½; hardbound. **$5.50**
Circle 8 on Coupon Below

Handbook of Broadcasting, 4th Ed.
Abbott and Rider. This non-technical handbook sets forth clearly and thoroughly the fundamentals of radio and television broadcasting. Over 100 illustrations amplify the text.
1957. 531 pp. **$9.50**
Circle 38 on Coupon Below

Electronic Musical Instruments
by Richard H. Dorf. Now in its third edition and sixth printing since its first appearance in 1954, this is considered the authority on electronic organs. 393 pages; 239 diagrams and photographs. **$10.00**
Circle 19 on Coupon Below

Intermodulation and Harmonic Distortion Handbook
by Howard M. Tremaine. A complete reference guidebook on audio signal intermodulation and harmonic distortion. 172 pages; 5½ x 8½; softbound. **$4.25**
Circle 9 on Coupon Below

Solid-State Electronics
Hibberd. A Basic Course for Engineers and Technicians. An extremely practical reference book for anyone who wants to acquire a good but general understanding of semi-conductor principles. Features questions to answer, problems to solve. 1968. 169 pp. **$9.95**
Circle 31 on Coupon Below

Practical PA Guidebook: How to Install, Operate and Service Public Address Systems
by Norman H. Crowhurst. 1967. This book gives all the basics needed to become a successful PA operator, in any situation where the reinforcement, relay, or distribution of sound can provide a service. 136 pages; 6 x 9; illus.; softbound. **$4.50**
Circle 15 on Coupon Below

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

FOR SALE

MODULES FOR ALL TYPES of applications. Discrete to thick film IC devices. Preamplifiers to power amplifiers. New Catalog with schematics, $1.00. **PM Electronics, Inc., Box 46204, Seattle, Washington 98146.**

SCULLY TAPE RECORDERS — one to twenty-four track and model 270 auto players, many models in stock for immediate delivery. SCULLY LATHES — Previously owned and rebuilt. Variable or automatic pitch. Complete cutting systems with Westrex heads. MIXING CONSOLES — Custom designed using Wiegand Audio Lab modules. From $7,000.00. **Wiegand Audio Laboratories, 3402 Windsor Road, Wall, New Jersey 07719. Phone: 201 681-6443.**

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.**

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.**

PULTEC PROGRAM EQUALIZER model EQH 3—$150.00. Pultec high pass low pass (program) filter model HLF 1—$75.00. **Phone 201 567-0500 weekdays 9:00 am to 5:00 pm.**

SOUND LEVEL METERS, General Radio 1551C—$285 each. New, never used. F.O.B. **S. Mieth, P.O. Box 456, Santa Ynez, Cal. 93460.**

COMPLETE TAPE DUPLICATING SYSTEM. Convertible 8-track/4-track. One inch master and bin with ten 1/4-in. slaves. Reliable, trouble-free operation. Some support equipment available. Greatly reduced price. **P.O. Box 65856 Los Angeles, Calif. 90065 Stn. 967**

UNIVERSAL OPAMP CARD. 3 x 3 epoxy board fits ten-pin connector. Input overload/output-shorting/power-supply-reversal protection. RIAA, NAB, or flat response characteristics. Wired flat. Gain externally adjustable 20 to 60 dB. Satisfaction guaranteed. $29.95. **Production Designs, 13828 Peyton Drive, Dallas, Texas 75204.**

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.**

WANTED

16-INCH DISCS WANTED containing old radio programs, comedy, drama, childrens shows, news, mystery, etc., or taped copies of the discs. **Don Maris, 1926 Cherokee Lane, Norman, Oklahoma 73069.**

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.**

PROFESSIONAL SOUND SYSTEMS ENGINEER. Must be knowledgeable in communications audio field, major installations, and systems equalization. Contact **Bernard Johnston, DuKane Corp., St. Charles, Illinois 60174. Phone (312) 584-2300.**

MATURE CHIEF ENGINEER with background in broadcasting. We are a 5000W daytime and 500W DA-night. We operate via remote control 24 hours. If you are a qualified broadcast engineer, self-starter, and take pride in your work, let's get together. We offer five figures and a future to the right man. No phone calls please. Send complete resume and photo to: **Bernie Barker, v.-p. general manager, WDAK Radio, Box 1640, Columbus, Georgia 31902.**

PEOPLE, PLACES, HAPPENINGS

• **The Society of Motion Picture and Television Engineers** will hold its 109th Technical Conference and Equipment Exhibit April 25-30 at the Century Plaza Hotel in Los Angeles. The Conference is expected to draw over 3,000 scientists, engineers and executives from the U.S., Canada, and abroad to participate in the five days of technical papers presentations and to view the latest in motion-picture and television equipment.

Two long sessions will be devoted to Television Systems on Tuesday and Wednesday afternoons, April 27 and 28. A total of 15 papers will be presented in the two sessions.

The area of television also includes the use of motion-picture film for television transmission. New techniques for transferring color video-tape to film have been developing over the past few years. In a paper by K. G. Lisk and C. H. Evans, Research Laboratories, Eastman Kodak Co., Rochester, N.Y., entitled, "Color Television Film Recording From a Shadow-Mask Picture Tube," developments in this process are described. This method is the direct photography on color film of a color TV picture played back from a tape and displayed on a shadow-mask tube of the same type used in a home receiver. Problems of this type of transcription are discussed with suggestions for improving the overall quality.

In the development of more sophisticated color television equipment new improvements are being made every day. "A Systems Approach to Linear Integrated Circuits for Color Television Facilities," by Hiroshi Naitoh and Yutaka Itoh, Tokyo Broadcasting System, Inc., Tokyo, describes an attempt to standardize basic color TV circuits in order to resolve cost problems and establish practical performance specifications without causing deterioration of picture quality. Standard equipment redesigned with this new circuitry will reduce design cost and make operation and maintenance much simpler and trouble free.

SMPTE Conferences are held twice yearly in major cities in the U.S. and Canada providing a forum for the exchange of new ideas and techniques as well as the introduction of new equipment in the motion-picture, television and photographic instrumentation fields.

• We note with deep sadness the passing of Sherman M. Fairchild, one of the truly great pioneers in the audio industry. His tremendous energy and technical skill will be long remembered. Sherman Fairchild was the son of George W. Fairchild, the first president and chairman of IBM and from his earliest days he was enthused over mechanical devices, many of them collected in visits to the IBM factory. After attending Harvard in 1915 where he designed the forerunner of the modern news flash camera, he left for Arizona for treatment of incipient tuberculosis. He attended the University of Arizona and later Columbia, but never bothered to get a degree.

Not long after, he built his famous aerial camera. Left a reputed $2,000,000 by his father in 1924, he started Fairchild Aviation which grew to a $6,000,000 business in just a few years. In 1936 he formed Fairchild Engine and Aircraft Corp. and turned the older Aviation Company into Fairchild Camera. In 1957, he backed several young scientists then working for Beckman Instruments after several companies had turned down their request for financial backing. He set them up as Fairchild Semi-conductor Corp., a division of Fairchild Camera. Their success in the development of transistors is now legendary.

Fairchild Recording Equipment Co. (now known as Fairchild Sound) was a company dear to his heart. He started it after Fairchild Camera decided to drop their audio products shortly after World War 2.

Fairchild Recording quickly became renowned for advanced professional products, many of them much too early to achieve the recognition they should have deserved. Several firsts were the gear driven turntables, variable pitch recording lathes, the pic-sync tape recorder (some of which are still in use today and considered by many the finest professional equipment at any price), the famed moving-coil cartridges, the turret head arm, the first stereo cartridge, one of the early stereo cutting systems, and the early use of cadmium-sulphide cells in audio control circuits. Today Fairchild Sound has grown to a top position in the audio market with a wide ranging product line for the broadcast and recording industry.

Fairchild was a man of amazing versatility. Among his many other accomplishments was the design of the first U.S. plane with an enclosed cabin for pilot and passengers, an intense interest in architecture (he helped design his New York town house), his cooking skill (he studied at Paris' Cordon Bleu cooking school), tennis and jazz (he played a fair piano and recorded many of the famous names at his home where he maintained a modern recording studio).

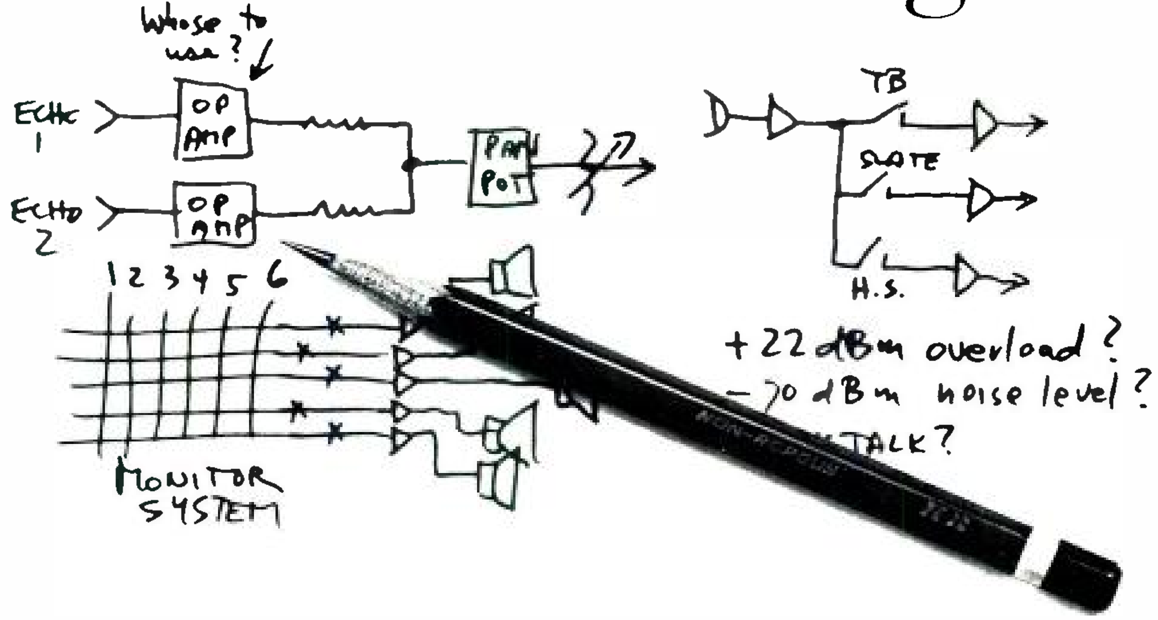
His life's work can best be summarized by a statement he made some years ago—"My sole objective is not to make money but to do something that is a substantial improvement over what has been done before."

Having once worked directly for Sherman Fairchild and having experienced his probing mind and devotion to our industry we feel a deep personal loss in his passing—he was truly a giant.

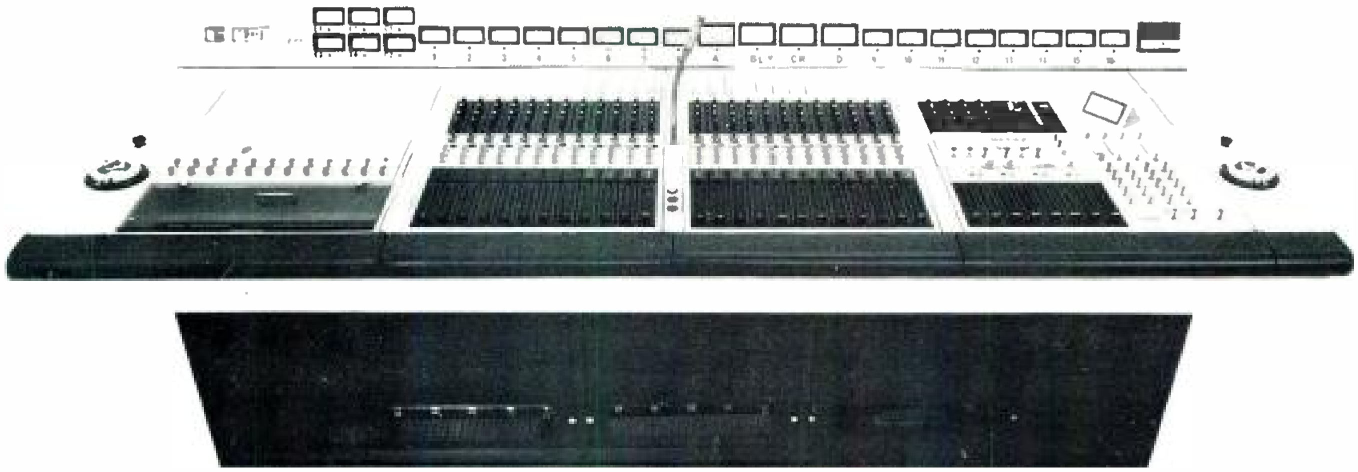
RB

MAXELL
F-20

The studio console you're now dreaming about



can be delivered next week.



Stop doodling and start making money. We have your 16-track dream board in stock. Ready to go to work for you, just as soon as you give the word. (Five 8-track boards are already in service from coast-to-coast.)

This remarkable unit was designed and built for us by Neumann, the finest name in audio equipment. It has every important feature you'd specify if you were having it custom-made. Plus a few even the most designing engineer might overlook. Consider these features:

Twenty-four inputs, and sixteen output channels, with mix-down to 8-track, quad, and stereo simultaneously.

Plus: 16 pan-pots, 2 cue and 4 echo channels, 4 remote echo controls, 4 remote tape controls, talk-back limiter, day-date-time digital clock, digital electronic stop timer, U-79 Correlation Meter (Compatibility Guide), full 312 jack patch bay, and phantom powering for your Neumann FET-80 microphones.

It even has two cigarette lighters and ash trays. Unless you insist on using the faders for that. They don't mind.

The price? No, it's not inexpensive. This console is built, after all, to the same standards as Neumann microphones. For continuous service-free performance. So, call us today. You'll sleep better tomorrow if you do.

GOTHAM
AUDIO CORPORATION

2 West 46th Street, New York, NY 10036 (212) CO 5-4111
1710 N. LaBrea Ave., Hollywood, CA 90046 (213) 874-4444

NEUMANN

Circle 12 on Reader Service Card