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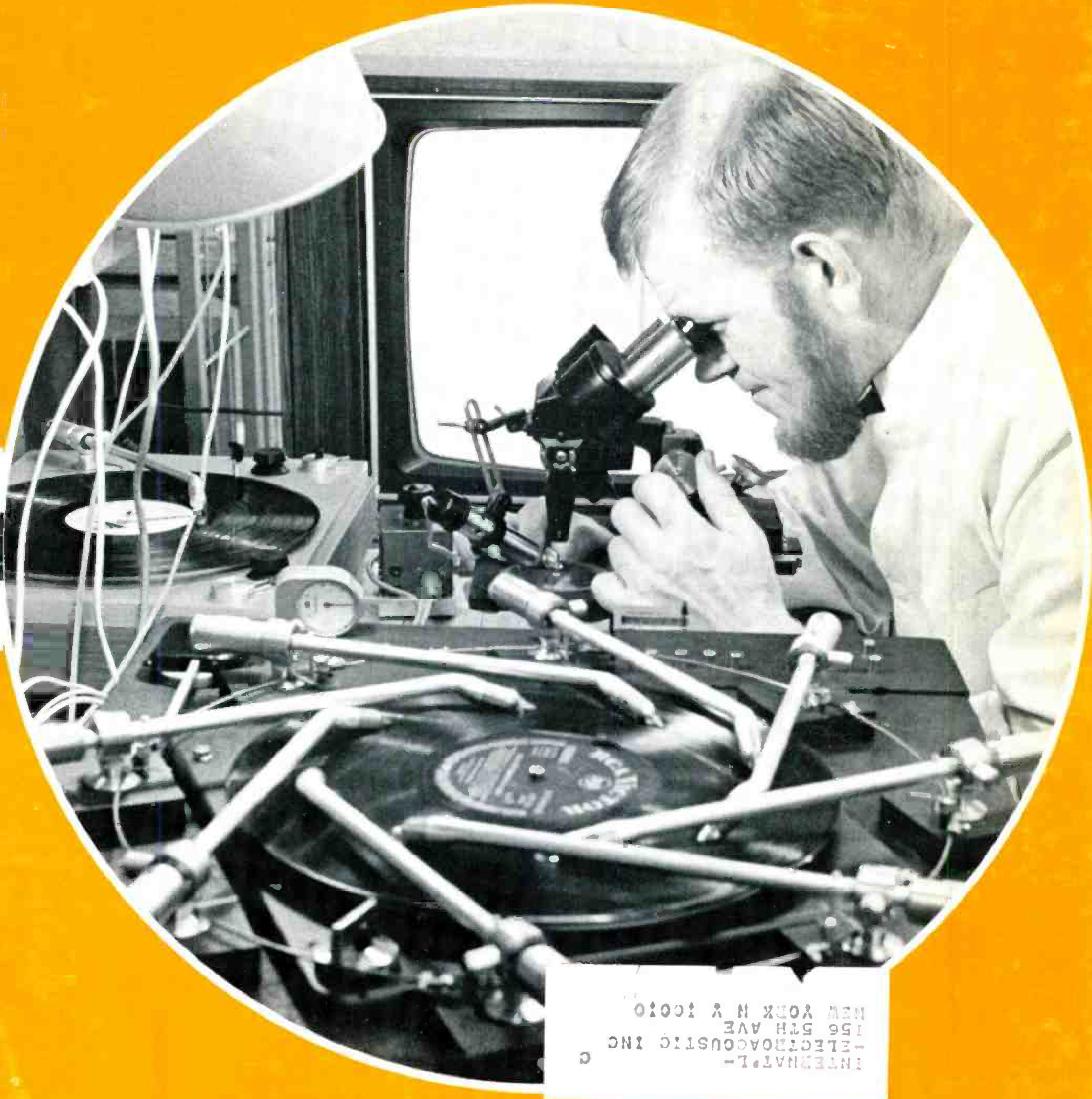
THE SOUND ENGINEERING MAGAZINE

DECEMBER 1971 $1.00



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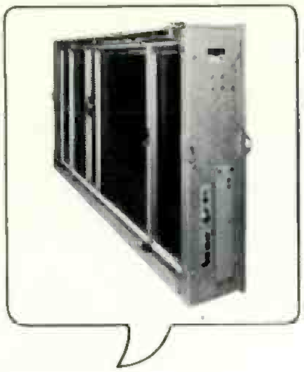

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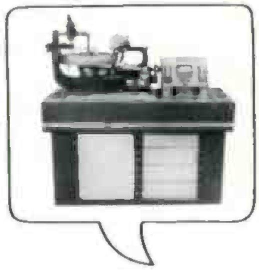

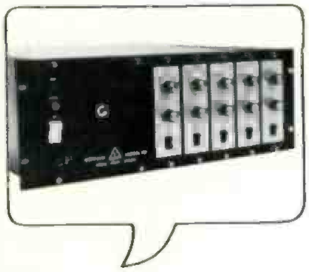

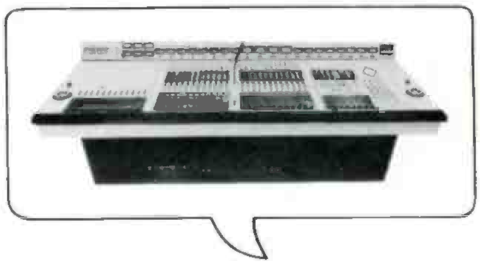

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THE SOUND ENGINEERING MAGAZINE

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George Alexandrovich

THE AUDIO ENGINEER'S HANDBOOK

● We, in the professional audio field, hear about level or gain control almost every day; be it compression, limiting, combining several audio channels or controlling the balance of four quadraphonic channels, or a miriad of other similar cases.

However, the most obvious immediate problems escape our attention. For instance, a few weeks ago I happened to travel on the Long Island Railroad in the brand new electric car that was only a few months old. As the train was approaching the station, the conductor made an announcement over the p.a. system. I couldn't hear the words or understand what he said because the sound coming from the speakers was more like a shotgun blast creating pain in my ears. It was obvious that the p.a. system was designed without any compromise in the amount of power it could deliver, or amount of amplification it could produce. The most important thing was missing—the automatic gain control to control the amplification and output level of the system. During the pauses, one could hear with great clarity the background noises, such as the compressor pump, or conversations of people who were standing nearby. After I disembarked that train, I had to take the subway. Almost the identical thing happened there, except that at the station, while the train was standing still and noise was comparatively low, announcements were clear and loud. As soon as the train would start moving and another announcement was made, it was almost impossible to understand what it was all about, not only because words were masked by the train noises picked up by the microphone, but also ambient noise in the subway car was so high that the level at which announcement was made was insufficient to be heard.

As we strive to perfect the ways of sound reproduction in the world's most sophisticated systems, it seems only natural that we should be solving such comparatively simple problems as designing sound systems for trains, subways, airports, theaters, churches and schools, etc. and contribute to solving the problem of noise pollution.

I mentioned the words *ambient noise*. Actually it is the key to the problem. Announcements in trains, in airports, in airplanes should be made at levels automatically controlled according to the ambient noise. As the surrounding noise level increases (such as at the airports when jets taxi in or out), the level of the p.a. system should increase accordingly. The same waiting area may be very quiet in the late hours when no planes are departing or arriving, and the announcements can be made at soft and pleasant levels that do not raise ones blood pressure.

Actually there are at least two systems now being manufactured that do this. But the application of the idea seems to be moving at a snail's pace. Partially this may be because only very experienced sound contractors are willing to tackle the solution of the problem, and partially that the equipment to do the job is too costly. But I am sure that if we put our minds to it, every p.a. system (sound reinforcement system?) could be supplied with the circuit which would control gain according to the ambient noise.

The two systems mentioned above

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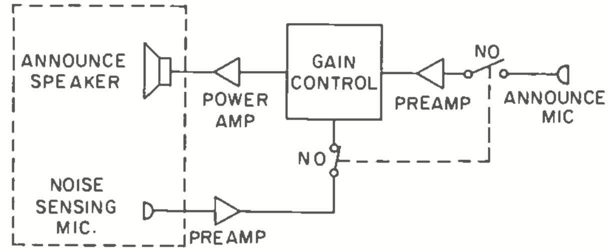


Figure 1. A noise sensing circuit.

operate on the following principles: one system uses step switching of levels, with holding circuits functioning indefinitely as long as announcement is being made. The other unit works with l.d.r.'s using solid-state control of a light shining on the cell. Cell resistance changes, affecting loss of a pad, thereby changing gain. A capacitor in the base of the driver transistor for the light source provides slow rate of change of bias, thereby producing slow changes in resistance of the l.d.r. cell. When the bleeder resistor is disconnected from the capacitor, a charge developed on the electrolytic keeps the bias almost unchanged for a few minutes, allowing ample time for announcement. The sensing circuit is disconnected at that time also.

I had the opportunity to work on such a unit, and criteria which had to be met were as follows: The device had to perform the following functions; control the level of the system with change in ambient noise over the range of at least 20 dB. The unit had to take signals from the noise-sensing microphone(s), amplify them and use them to change the gain of the announce mic circuit. The relationship between the change of ambient noise and announce levels had to be linear. For every dB of noise, increased level of the announcement had to be raised one dB.

Change of level had to be gradual, responding to average noise level rather than individual bursts or peaks. As the noise receded, gain of the system was to allow to change slowly over the period of several seconds.

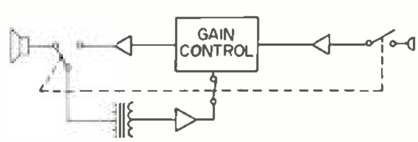
Noise sensing systems had to be insensitive to extremely low- and high-frequency noises which normally don't interfere with hearing the announcements. Because of this, sensing microphones can be lower quality units, lacking in low- or high-frequency response.

Two types of circuits were needed. One which would operate all the time and where sensing microphone(s) were located outside the announcement area (in airports outdoors, where jets taxi) and a system which had sensing mics next to the announce speakers and required temporary shut-off of sensing when the announcement was being made. This required a special freeze circuit which would hold the gain constant over the period of a couple of minutes at the level just preceding the announcement, and which would at the instant the announcement was over, commence sensing again.

In less sophisticated systems, announce speakers are being used for noise sensing, and for announcement switched to the output of the p.a. amplifier. This circuit cannot be used in systems with continuous sensing (for instance, when the level of the background music in a restaurant has to be controlled). In this case, directional sensing mics have to be used, so that they pick up the din of the crowd and not the music, otherwise level of the system may start increasing until it reaches maximum. It is nothing else but the positive feedback. If the action is allowed to go uncontrolled, sound will bloom until it will drive you out of the room. When the noise sensing is done outside the announce area, then the problem of level control is simple. Take trains, for example. All you have to sense is the noise of the train, since it is a domineering factor in setting the gain of the announcements. True that the train noise heard outside the cars is much greater than the noise heard inside the car, but then again, the rate of change in p.a. level can be adjusted to produce the desired effect in tracking changes in ambient noise.

I want to now talk about the levels involved in such systems. Let us start with noise sensing. Noise levels can vary from the quiet of a great cathedral to the ear-piercing noise of the air hammer or steam turbine. In the average installation, ambient noise will produce sound pressure resulting in mic levels of —80 to —90 dBm, using an average dynamic mic. It means that you will need preamplification on the

Figure 2. A method of using the announce speaker for noise sensing.



shhhh!

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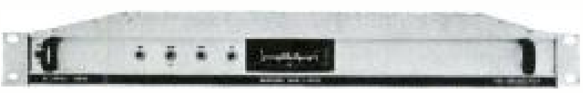

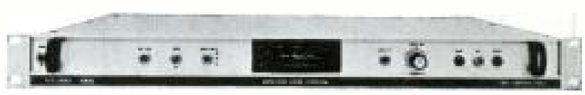

This is the team that quietly goes to work to produce a perfect sounding program. Audimax eliminates distortions like thumping, audio holes and the "swish-up" of background noises. Volumax prevents overmodulation and limits program peaks, permitting broadcasters to achieve maximum power from each watt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

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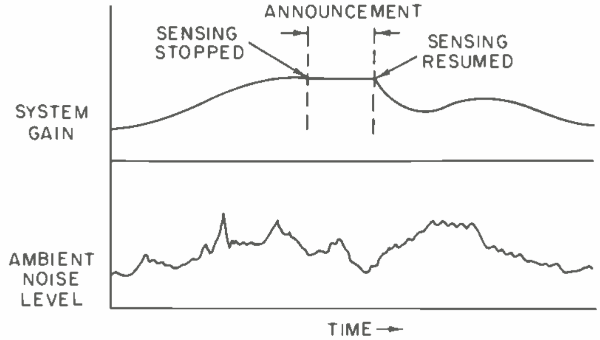


Figure 3. Typical system gain; noise against time.

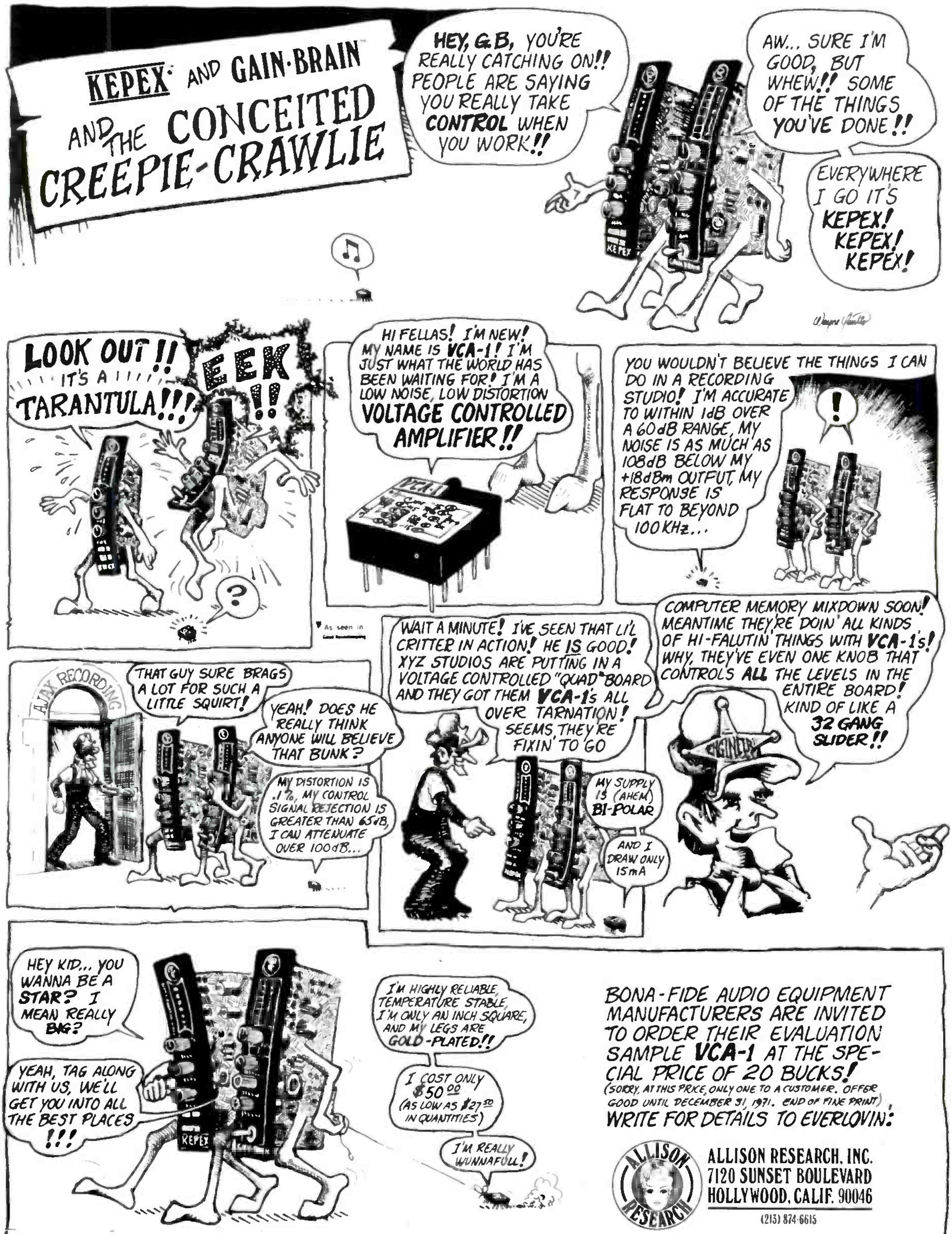
order of not less than 80 to 90 dB before line levels can be achieved. Luckily these signals are only used for the d.c. control of the gain-riding circuits. Restricting frequency range further reduces troublesome possible transients. When speakers are used for sensing, it is important that the resonant frequency of the speaker be filtered out, otherwise signals with frequencies near or at resonance of the speaker may produce incorrect gain riding.

More complications occur when the same system is called upon to perform more functions, like between the announcements it is supposed to supply background music, yet when the announcements are made it should adjust gain according to the ambient noise level. Several solutions in this case are possible. Either separate sets of speakers are used for background music, or separate sensing mics. Part of the background signal with inverse phase is fed into the mic channel, so that the effect of the music is minimized in setting the gain of the system. I have heard about experiments conducted with the speakers; using them simultaneously for music and sensing. It would seem that with lots of phase shifts it would be a hard task to perform—but if you put your mind to it, almost every problem can be solved.

In the past few months, I have been talking more about communication problems than problems of recording and sound processing. Well, I think that these problems are more pressing. I am not saying that we should stop doing what we have been working on, but I would like to see abandoned fields get attention. How many of us realize that audio *is* the most important means of communications. Television cannot survive without sound, nor telephone, nor movies. The last AES Show was somewhat of a disappointment to me. It seems that it could be called the Studio Equipment Show. The best engineering talents, which should dictate audio standards to *all* facets of communications, have concentrated on sixteen-track recording. What a pity! ■

moving

Have you sent us a change-of-address notice? It takes time for us to change your plate so let us know well in advance of your move. Be sure to send us the complete new address *as well* as *your old address.* Include both zip numbers. Keep **db** coming without interruption!



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John M. Woram

THE SYNC TRACK

The 41st AES Convention

● Here in Fun City, another AES convention has passed. By the time these words reach print, the exhibits will have been gone for well over a month, and the AES Picture Gallery (page 17) will be on its way to becoming nostalgia.

Around the exhibits, there was again heard some talk of limiting the convention to a once-a-year affair. Admittedly, it's not an easy matter to transport heavy consoles back and forth across the country twice a year. In fact, there are now some companies that only exhibit in California, others only in New York.

Yet, there are certainly enough technical papers offered to justify two or *more* conventions a year. In fact, both New York and California have had to run concurrent sessions in two meeting rooms, just to get all the papers presented within the convention week. A single annual convention would *really* present a logistical problem in this area.

As for exhibits, the AES and the exhibiting manufacturers provide a valuable service to the audio industry by bringing together—on both coasts—the latest equipment. Many AES members have neither the time nor the money to journey cross country, and since there is such a concentration of audio activity in both the California and New York areas, the present arrangement allows both areas an annual look at what's going on.

Of course, it would be nice if the society could find a more suitable hotel in New York City. The virtues of the present site—such as many another midtown landmark, both of flesh and of stone—are nebulous at best. Yet, hotels that are less depressing are either incredibly expensive, uninterested, or incapable of handling the requirements of the convention. Until a more suitable location can be discovered, the present site at least has sufficient exhibit space.

NEW PRODUCTS SHOWN

Many of the tape recorder manufacturers had interesting new products to show. At last, the Studer A-80 sixteen-track recorder was displayed. The unit uses separate plug-in cards for the eight-track head assembly. At this time, a remote control unit is not available.

Ampex displayed a new sixteen-track machine that is about the same size as the 3M deck. But to me, their most interesting new product is the remote read-out and search unit for the MM-1000 series deck. This one reads in real time, which is a tremendous advantage in any studio situation. It's very difficult to get an artist or producer to relate to the number of revolutions of the supply reel. And, one revolution may mean anything from about one foot to one yard of tape, depending on the amount of tape remaining on the supply reel. Certainly, a revolutions counter is better than no counter at all, but a real-time indication is preferable. The MM-1000 tape path already contains a mechanical footage counter with hours/minutes/seconds read-out, so the remote unit involves no major design changes. Some other machines have no existing slippage-free rollers, so a real-time device would require at least a little redesigning of the tape path—no easy matter.

The problem of real-time readout might be simplified if some tape manufacturer could be persuaded to print some form of stripe on the tape backing which could then be read by an external optical unit. Since Agfa tape already contains printing on the backing, I presume there are no unsurmountable obstacles to printing a pattern that could be read by a light-sensitive device.

Coming back to tape transports, 3M brought along a prototype that outwardly resembled their current sixteen-track deck. However, it was remarkably faster handling, changing direction and stopping in a fraction of the time required by most other machines. It seemed to be about the fastest, most responsive machine I've seen, with the exception of a custom built transport constructed by Claude Karczmer at Vanguard Records. (More about this later.)

Another feature of the 3M prototype is a built in Dolby unit — or rather sixteen of them. This should certainly be welcome news to all those who spend so much time plugging and unplugging Dolbys, and wondering if they've properly switched the terminations off (or was it on?)

And, on the subject of noise reduction, dbx and Burwen Laboratories both had new noise reduction units to show.

The entry of these systems to the audio market invites a whole series of questions, such as: What about compatibility between contending systems? What is the optimum amount of noise reduction? Do new developments in tape manufacture minimize the importance of noise reduction systems? and so on.

As an aside, I mentioned the dbx extended range meter in the August issue. Now that we've been using it for some months, I wonder how we ever got along without it. How nice it is to occasionally find something that is both inexpensive and useful.

Eventide Clock Works, Inc. (!) is a convention newcomer. Many of their products are still in the advanced development stage, but show considerable promise as useful gadgets for the studio looking for a competitive edge. Among other little goodies soon to be available (I hope) are a phasing unit, a digital delay line, a stereo transmitter, and a rate pitch changer.

AKG displayed their new reverberation unit. Earlier this year, I heard the unit demonstrated, and found it compared favorably with other electro-mechanical units. Where space is at a premium, it has the advantage of being reasonably small, and since it is fairly light weight, I imagine it could be readily moved around—even taken on remotes.

Allison Research's limiter or 'Gain Brain' and Allison's Kool-Aid shared the booth with the well-known Kepex. The Gain Brain is a combination peak and r.m.s. limiter. I've been fooling around with one for a few months now, and have found it works very well in conjunction with the Kepex, especially with inexperienced bass players.

Of course, the leading console manufacturers were there, with their latest developments. I would still like to see a little more human engineering designed into consoles, but I suppose I'm some sort of fanatic or something. In addition to knobs that point the way, and legends that are not obscured from view unless the engineer is standing on top of the console, I'd like to see (among other things) two echo returns for every echo send, or at least more returns than sends; flexible panning facility, echo equalization, separate speaker volume controls, ganged faders for two- (or four-) track mixdowns, cue systems with separate volume controls for each track in each cue line, meaningful color coding, and so on, and on, and on. I almost forgot—how about an oscillator that is continuously variable between at least 10 kHz and 1 kHz so you *really* know what's happening? Yes, I know that oscillators are available separately, but how about building one into the console so it will be there when you want it?

Admirers of the Neumann U-47 will be interested to know that the transistor version of that famous microphone will be available momentarily from Gotham. And Electro-Voice now has a line of inexpensive electret condenser microphones available.

Other interesting microphones shown were the AKG series of modular microphones and the recently introduced line of Beyer mics. The AKG system features a collection of capsules, bodies, extensions, angles, and whatnot, all of which can be interchanged. For example, a right angle assembly may be inserted between the capsule and the microphone body, or an omni assembly can be quickly replaced with a cardioid, and so on.

Last but not least, and with apologies to all the unmentioned exhibitors, International Telecomm showed their parametric stereo equalizer which appeared to be quite a flexible unit, and might turn out to be particularly suitable for doctoring up a problem track.

But now its time to start thinking about the 42nd AES convention to be held in Los Angeles May 2-5, 1972. ■

Technically, our new SQ four-channel system has 4 basic advantages.

Sony SQ.

A new stereo/quadraphonic system, delivers four distinct sound channels from a compatible SQ record.

It also offers four distinct advantages over all the other four-channel "matrix" systems.

Advantage #1:
Greatest stereo separation, front and rear.

Your present stereo system probably can maintain 40 db or so of separation between left and right channels. Maintaining this full left-to-right separation, in both the front *and* rear pairs of channels, is one of the major achievements of the SQ system.

Advantage #2:
Simple logic that lets soloists stay soloists.

When a single instrument is playing, all you want to hear is that instrument, even in four-channel. A pure matrix decoder—even the matrix at the heart of SQ—can't reproduce a solo instrument without a softer, phantom soloist in other channels. But by adding a logic circuit, these phantom signals can be diminished or eliminated, sharpening your sense of the soloist's position.

So far, though, only Sony's SQD-1000 and SQA-200 decoders have this logic enhancement. Because SQ's unique encoding (which shows up on records as a double-helical modulation of the groove) makes simple logic circuits practical.

Advantage #3:
Total omnidirectional fidelity.

A musician plays no softer when he's behind you or to one side. With SQ he doesn't sound as if he did. No matter where in the 360° quadraphonic circle the musician sits, he will be heard at exactly the same volume as if he were sitting in front of you. And that's true whether you're listening to the SQ record in four-channel or just playing it on a stereo system without an SQ decoder.

Advantage #4:
Equipment by Sony.

Sony offers you a choice of two SQ adapters. For the more demanding, there's a new SQD-1000 decoder. Its logic circuit enhances front-back separation by up to 6 db, so that front-center soloists (or rear ones, for that matter), stand out more clearly. The SQD-1000 lets you listen to four-channel sound from SQ records, or to discrete four-channel tapes on auxiliary players. It also lets you listen to normal stereo, or to stereo broadcasts and recordings enhanced with SQ ambience. Just plug the SQD-1000 into your tape monitor jacks (the SQD-1000 has its own), and add your choice of rear-channel amplifier and speakers.

If you want to get into SQ with a more modest investment, add Sony's new SQA-200 SQ decoder/amplifier to your system. It has all the SQD-1000's features (except the four-channel master volume control). But because the SQA-200 has a stereo amplifier built in, it saves you the expense of an extra amplifier for your rear channels.

Hear SQ at your Sony dealer. Or write Sony Corporation of America, 47-47 Van Dam Street, Long Island City, N.Y. 11101.

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Musically, it's starting out with 52.

Columbia Records

Popular

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Blood, Sweat and Tears II
Johnny Cash at San Quentin
Chase
Ray Conniff, *Love Story*
Al Cooper, Mike Bloomfield & Steve Stills, *Supersession*
Miles Davis, *Bitches Brew*
Bob Dylan, *Nashville Skyline*
Percy Faith, *Romeo and Juliet*
Funny Girl, Original Sound Track
Janis Joplin, *Pearl*
Kris Kristofferson, *Silver-Tongued Devil and I*
Johnny Mathis, *You've Got a Friend*
Jim Nabors, *Help Me Make It Through The Night*
No, No, Nanette, Original Cast
Poco, Deliverin'
Ray Price, *For the Good Times*
Raiders, *Indian Reservation*
Santana, *Abraxas*
Sly and the Family Stone, *Greatest Hits*
Ray Stevens, *Greatest Hits*
Barbara Streisand, *Stoney End*
Ten Years After, *A Space in Time*
Andy Williams, *Love Story*
Tammy Wynette, *We Sure Can Love Each Other*

Classical

Bach, *Switched-On Bach* (Carlos)
Bernstein, *Mass* (Bernstein, Original Kennedy Center Cast)
R. Strauss, *Also Sprach Zarathustra* (Bernstein, N.Y. Philharmonic)
Morton Subotnick, *Touch*
Verdi, *Requiem* (Bernstein, Arroyo, Veasey, Domingo, Raimondi, London Symphony)

Vanguard Records

Popular

Joan Baez, *Blessed are...*
Larry Coryell, *At the Village Gate*
Country Joe and the Fish, *From Haight-Ashbury to Woodstock (2 LP)*
Buffy Sainte-Marie, *Moonshot*

Classical

"P.D.Q. Bach," *The Stoned Guest* (Schickele)
Berlioz, *Requiem* (Abravanel, Utah)
Handel's *Messiah* (Price, Minton, Young, Diaz, Somary, English Chamber Orch.)
Handel, *Messiah highlights*
Mahler, *Symphony No. 3* (Abravanel, Utah)
Mozart, *Divertimenti* K287/138 (Blum, English Chamber Orch).
Tchaikovsky *Symphony No. 4* (Stokowski, American Symphony)
Tchaikovsky, *Serenade Op. 48;* Prokofiev, *Classical Symphony;* Arensky, *Variations* (Somary, English Chamber Orch.)
The Virtuoso Trumpet of Martin Berinbaum (Somary, English Chamber Orch).

Ampex Records

Popular

Anita Kerr Singers *Grow to Know Me*
Anita Kerr Singers with Royal Philharmonic, *A Christmas Story*
Bob Hinkle, *Ollie Moggus*
Melting Pot, *Fire Burn and Cauldron Bubble*
Mason Profit, *Last Night I had the Strangest Dream*
Purlie, Original Cast
Cris Williamson
Rome Philharmonic, *Classical Movie Themes*

four-channel record system

Norman H. Crowhurst

THEORY AND PRACTICE

● I'm resuming the discussion of types of motors used for drives in audio equipment, and we have come to the question of providing a smooth, wow-less and flutterless, constant-speed drive for recording and playback, both of tape and disc recordings. For d.c. operated machines, such as portable recorders, a well-built permanent-magnet motor, supplied with constant d.c. voltage will maintain constant speed.

However, a d.c. motor will have varying torque every time a brush crosses the insulation in the commutator, so the current path through the armature is changed. The average torque and speed may be held constant, but this fluctuation many times during each revolution can cause flutter.

Consequently, the capstan requires sufficient inertia (rotating mass) to absorb all fluctuations of torque while keeping speed close enough to constant as to represent a satisfactorily low order of flutter.

Holding d.c. constant, so speed is reliably the same at all times, may not be easy when batteries are used for power. A relatively simple remedy for this is the use of an inexpensive voltage regulator, such as a zener diode, to hold motor voltage constant, as battery voltage varies.

Flutter and wow can be caused by mechanical effects as well as electrical variations. This means that the drive from the motor shaft to the point that drives the tape or disc surface needs consideration. Various types of drive can be used: direct (the capstan or turntable is on the same shaft as the motor), belt, or gear. Variations of the belt might use chain, while use of a jockey pulley, with no belt may be considered as a variety of gear.

We will not discuss the deriving of correct speed ratios, which is a relatively simple arithmetical calculation, but the fact that such ratios are neccessary, because motors may not turn at the same speed as required of the turntable or capstan, means that a drive using one or other form of conversion must be used. The alternative is to design the machine so the motor does turn at the same speed as the turntable or capstan.

For a turntable drive, this means the shaft must rotate at one of the standard speeds, given in r.p.m.: 78, 45, 33⅓ or 16⅔. Back in the days when one speed was standard, so that machines were commonly made with only one speed, one approach was to use a specially-designed synchronous motor. Operating on 60-Hz supply, the synchronous speed of a 2-pole motor is 3600 r.p.m.

Synchronous speed is inversely proportional to the number of poles, which must be even, because every north must have a corresponding south (and *vice versa*, I should add, to avoid being accused of geographical prejudice). So a motor built with 92 poles will run at 78.2 r.p.m. If you want 33⅓ r.p.m., the motor will need 216 poles, while 45 r.p.m. calls for 160 poles.

A single such motor may be feasible, but to make a motor that can be switched from 92 to 160 to 216 poles makes matters somewhat complicated, both electrically and mechanically. It is much simpler to design a good motor that will hold its speed constant to one value, and provide the different speeds by purely mechanical means, in the drive.

So most turntables get the required variety of speeds by using either a variable-diameter drum on the motor shaft, coupled to the turntable periphery (inside or outside, usually the former) by an intermediate jockey pully. The speed is determined by the diameter of the drum section on the motor shaft against which the jockey pulley engages.

The jockey pulley, which is usually the smoothest possible wheel with a rubber periphery, merely serves to transfer the peripheral speed of the portion of motor shaft against which the rubber engages to the periphery of the turntable. Smoothness of speed (freedom from flutter and wow) depends on uniformity of the three elements that engage one another.

The alternative form of drive for turntables uses a similar variety of drum or pulley diameters with a belt to transfer the movement to the turntable periphery.

The reason gears are not used is that their transmission is essentially non-uniform as teeth engage and disengage, however perfectly the teeth of the gears are made. The same is true about chain drives. The transmission may be completely smooth, but it is not uniform, and thus introduces a speed flutter.

For driving capstans in tape recorders, a direct drive is more feasible and most modern recorders employ this form. Achieving acceptable flutter and wow elimination is then a matter of precision bearings, trueness of rotation, so that movement is perfectly uniform.

While a variety of electric motors can be used, the best is generally conceded to be the hysteresis synchronous, when the supply is a.c. The induction motors, discussed in the last issue, can achieve quite good constancy of speed, if the rotor resistance is low compared to its reactance. But still the degree of drag provided by the tape can vary the amount of slip that occurs—the amount by which the speed drops below a theoretical synchronous speed.

An induction motor never runs at synchronous speed: the torque to make it run necessitates that its running speed be slightly below synchronous. If it is a 4-pole motor, running on 60 Hz supply, the synchronous speed would be 1800 r.p.m. A good induction motor will probably run between 1700 and 1770 r.p.m.

A hysteresis - synchronous motor starts as an induction motor: the same torque characteristic possessed by the induction motor gets it up close to synchronous speed. Then the fact that a slow slip causes slugs of magnetic material on the rotor to slowly reverse their magnetization draws energy from the supply, an energy that alternately boosts and retards the speed, if the motor continues to run as an induction type.

As the rotor drops behind one condition of magnetization of the slugs representing a synchronous operation, the energy drawn by hysteresis pulls the speed up, tending to prevent the rotor from slipping out of synch. As it passes beyond the critical point, to where opposite magnetization would occur at synchronism, the rotor is retarded by the hysteresis energy.

The design of a good hysteresis synchronous motor is such that the forward pull keeps the motor in synch, once it has run up to speed, against considerable torque trying to pull it out. The more it drags behind, the more hysteresis energy is drawn from the supply to keep it in synch, until it reaches pull-out point, just like a regular synchronous motor.

Some tape recorders use one hysteresis synchronous motor as a central drive unit (or perhaps a good induction motor, as a second choice) with a slipping-belt system to provide take-up drive to the take-up spool. This is a cost-saving feature, as a rule, because one motor costs less than three (one for capstan and one for each spool). Fast forward and rewind is usually provided by using a different belt configuration and a system of clutches.

The more sophisticated drives use the constant-speed motor to drive only the capstan, and employ an extra motor or two (more usually the latter) to drive the spools. Many prefer d.c. motors for this purpose, because a simple way of changing speed, to go from normal takeup during record or playback, is to change the voltage applied to the motor(s).

The fast forward and rewind functions, on any recorder and, more particularly, regular tape-movement functions on a high-speed duplicator, require special care for tape handling during speed run-up and stopping. With the fast-forward and rewind functions, from one viewpoint it does not matter how fast speed builds up (or how slowly, providing it is not aggravatingly slow) or how long it takes to bring the tape to a stop.

From another viewpoint, however, rapid stopping is desirable, and equally rapid starting to complement it is almost equally desirable. If the run-up and stopping is too lethargic, finding (or stopping the tape at) a predetermined spot can be difficult. The inertia inherent in the slow speed change means that the spot you want will go by, often several times, before you finally stop the tape precisely where you want it to stop.

The problem with starting, and more particularly with stopping the tape with such a degree of suddenness is that of ensuring that it never snatches with danger of breaking the tape, and never allows the tape to run slack, so as to allow it to leave its assigned path and tangle.

This involves careful attention to both start-up and braking, so the combined action at each point is smooth but rapid. Change from drive to brake must be properly sequenced with split-second timing. If you have ever operated a recorder where the designer relied on the speed at which you press the button down with your finger, you will have discovered that hitting the button either too quickly or too slowly will cause trouble. Usually hitting it too quickly will snatch the tape, running the risk of breaking it, while hitting it too slowly will loose control, allowing it to spill and tangle.

To avoid this, the more sophisticated designers use electric or electronic controls that take timing out of the user's control. The control button (or whatever form it takes) merely operates a single contact that initiates the necessary sequence of actions. This sequence may be electromechanical, determined by solenoids or relays with carefully controlled delays in action, or it may be electrical or electronic.

Solid-state circuits may program the necessary changes. Viewed as better than mechanical braking by many designers is electrical braking. A motor is used of a design such that applying d.c. causes braking. Then application of the d.c. can easily be controlled by use of suitable large value capacitors that allow the d.c. applied to the motor to build up at the required rate.

With all these problems in control of tape, and maintaining constant speed, whether tape or disc, there is usually more than one approach to solving the problem. While many argue about which is best, it usually comes back to the simple fact that each is only as good as the method it uses is applied. Almost any system, by careful attention to detail, can be made to work well. And what may theoretically be the best method can "come unstuck" when sloppily applied. ■

Martin Dickstein

SOUND WITH IMAGES

What Is Resolution?

● Although it may be near the New Year season, that is not the kind of resolution we mean. What is *television resolution?* Perhaps that pins it down a bit closer.

According to the Second Edition of the huge New International Webster's Unabridged Dictionary, which I carry around all the time and just happen to have in my back pocket, the word is given many definitions for its various applications, but there is none given specifically for t.v. The closest definitions which are applicable come from the fields of optics and photography.

Under optics, the word is defined as "the art or property of rendering visible the separate parts of an object, or of distinguishing between different sources of light." Then it says "See optical resolving power."

The definition given under "Resolving Power, optics" is "the ability of an optical instrument to form distinguishable images of objects separated by small angular distances."

In another place, under "Optical resolving power", the definition is given as "The power of an optical system using light of one color to form images of two point sources which are close together so that the images may be recognized as distinct."

In the definition for resolution in the sub-division of Photog., it says "the ability of a film or plate to reproduce the fine detail of the optical image".

It looks like a combination of several of the definitions given would apply to television, but the technical definition is given by RETMA as the maximum number of black-and-white lines visible in one picture height.

Since the term is defined in terms of picture height, and the standard aspect ratio of our standard television systems is 4:3, it follows that the horizontal resolution would be the number of horizontal bars visible in one picture height or in ¾ of a picture width. Although the term is defined as lines per picture height, it is common in general usage to eliminate the last part of the term and just refer to resolution as *lines.*

The horizontal resolving capability of a camera or monitor is determined by the system bandwidth, active line time and the aspect ratio. It is, therefore, possible for a very slow-scan system to have a high resolution with long scan periods being used. If the bandwidth is increased, the resolution would improve.

For relating horizontal resolution to bandwidth and scan time, the relationship is given as horizontal resolution factor (HRF) $= 3/2$ T, where T is the active horizontal line time, or horizontal scan time minus the time for horizontal blanking. HRF is given here in lines per MHz. and in this equation for a system with a 4:3 aspect ratio.

From this, it can be determined that for the U.S. 525-line system of television transmission, with a standard bandwidth of 4.2 MHz, and an HRF of 80 lines/MHz, the nominal horizontal resolution is 340 lines. In Europe and elsewhere, the t.v. systems use 675 lines and 945 lines. For the former, an HRF of 59 lines with a bandwidth of 7.5 MHz gives a resolution of 440 lines. In the latter system, a resolution of 615 lines is achieved with a bandwidth of 15 MHz and a factor of 40.5 lines/MHz.

For vertical resolution, the definition is also given in terms of lines per picture height and refers to the maximum number of horizontal black-and-white lines that can be seen in one frame. This means the total number of lines minus the number of vertical blanking lines. For an interlaced system, a number called the Kell factor is introduced. This term is found experimentally and relates the total number of *active* lines in a frame with the number that can be resolved visually. The number is found to be 0.7. Thus, the actual vertical resolution of an interlaced system is the number of active lines per frame times this 0.7 factor.

To find the actual resolution of a 525-line system, therefore, it is only necessary to determine the active number of lines, subtract from the over-all total and multiply by the factor. If it is considered that each field contains a period of 20 blanking lines, there are, therefore, 40 total blanking lines in each frame (2 fields per frame). The result is a resolution of 340 lines, the same as the horizontal resolution found previously.

It should be indicated here that for random interlace systems the Kell factor does not apply. The resolution, then, would be simply the number of active lines per frame.

From the relationship given for vertical resolution, it can be seen that bandwidth has no effect on this capability of the system. Thus, in the situation of a slow-scan system, the vertical resolution would be improved by increasing the scan rate, but would be unaffected by increasing the bandwidth.

Measurement of the t.v. system horizontal resolution is performed by use of a special chart. Originally called RETMA charts, they are now referred to as EIA charts as they are published by the Electronic Industry Association. The various charts available in various shops may be slightly different, but the method of checking the resolution is the same and can be performed with any of the charts.

After the proper camera adjustments have been made for beam and target, etc., and normal lighting set up, the chart should be placed directly in front of the camera and perpendicular to the axis of the lens. The distance between the lens and the chart should be such that the infinity range of the lens must be used for the test.

If a zoom lens is used, it should now be zoomed until the chart just fills the viewfinder or monitor. This can be judged by the arrows (two on each side of the pattern) which should be at the extreme edges of the image or by the outside lines of the chart. By adjustment of the vertical or horizontal size of the image, the outside edges of the chart can be made to fit exactly the screen of the monitor. With the aspect ratio thus properly set, the resolution can now be read directly.

Just above and below the center of the chart, there are precisely spaced vertical lines. The widest spacing is marked with the number 200 and the narrowest, toward the center, with a 400. Another similar set of vertical lines is marked beginning with 400 and ending with the extremely fine spacing at the center marked with an 800. After adjusting the brightness and contrast for the best possible image, the resolution can be read at the point where the lines seem to lose their individuality and come together. If the marking at that point is 200, that is the resolution of the system. If the blurring occurs between two number markings, the resolution can be interpolated.

With the resolution at the center of the image now known, it is also

(continued on page 33)

NEW PRODUCTS AND SERVICES

Picture Gallery—N.Y. AES Convention

Herewith, the results of our camera's inquisition into the products and views seen at the Audio Engineering Society's Convention and Exhibition held recently in New York City. Space permits only highlights, and, at that, not all of them. We do have more interesting photos that will make their appearance next month.

If you wish information on any products shown, merely circle the appropriate reader service number on the post card at the rear of this issue and it will come directly from the manufacturer.

Views Around the Show

Special Highlights

Ampex and 3M both showed working prototypes of upcoming sixteen track recorders that will cost less but perform with more versatility than present models. Shown is the Ampex version.

Perhaps the busiest spot in the show was at the Neve booth. There a sixteen-track tape was providing signal so that a visitor could manipulate the board as if he were operating a live session. The tape was prepared by **db** associate editor John Woram (Vanguard Records) and he used Shure mics exclusively for it. The sixteen-track tape was also piped to the nearby Shure booth for continuous earphone monitoring there. For information on the Neve console, Circle 100 on Reader Service Card.

The newest ReVox has Dolby-B built in; also the range of Beyer mics and stands was displayed. Circle 90 on Reader Service Card.

Solid-state electronics and ingenious suspension are combined in this new delay system from AKG. Circle 99 on Reader Service Card.

Plug-in record-play modules are a feature of this sixteen track recorder from Auto-Tec. Circle 73 on Reader Service Card.

Melcor's Model 5001 is an all solid-state digital delay system with adjustability of 6 to 275 msec. Circle 93 on Reader Service Card.

Shure set up their mixers so that visitors could monitor a sixteen-track tape input fed through them. Circle 80 on Reader Service Card.

Eventide Clock Works, a new company, showed a prototype of a digital rate and pitch control. Circle 51 on Reader Service Card.

Circle 24 on Reader Service Card →

ROBINS INDUSTRIES CORP. SUBSIDIARY
15-58 127TH ST. FLUSHING, NEW YORK 11356 212 445-7200

REVERBERTRON MODEL 659A

The FAIRCHILD REVERBERTRON is a dynamic reverberation system designed to enhance broadcast and recording studio sound. Not only is a more pleasing sound obtained with the use of the REVERBERTRON, but a side effect is the production of sound that is apparently louder and "livelier" than a non-reverberated sound. The result is extremely natural sounding, and possesses the quality of good acoustical reverb chambers.

The REVERBERATION uses six electro-mechanical delay lines, each tuned differently to produce the natural reverberant effect. These mechanical devices are isolated to prevent building rumble and environmental noise pickup. Change in reverberation periods is obtained through the use of a compressor, which emphasizes decaying reverberation, effectively prolonging the effect.

Now smaller in size, several of these flexible reverberation systems can be stacked in the space previously occupied by a single system. The required rack space is only 7", or it can be carried into the field in one or two enclosures. The exclusive knob controlled "system lock" feature, which prevents vibration and damage in transit, and its low weight enhances its field use capability.

FEATURES

- Improved S/N (65 db).
- Lower input levels - 30 dbm.
- Full range equalization (bass, midrange and control presence peak selection, high and roll off).
- Instant selection of 3 decay times.
- Metering of all signals in the system.
- Local and remote selection of 3 degrees of reverberation (dry, - premix 1, premix 2).
- Selector switch for dry signal, fully reverberated, partially & remotely selectable.
- Continuous mix control.
- All electronics on plug-in P.C. boards for easy access and maintenance.
- Transformer isolated input and output 600 ohm or 150 balanced or unbalanced.
- Only 3-1/2" of rack space for electronics - 7" for the complete system.
- "System lock" for transit.

Patent #3436674

PERFORMANCE SPECIFICATIONS

Input Level:	-30 dbm or higher.
Input Impedance:	600 ohm balanced or unbalanced.
Output Handling Capability:	+18 dbm.
Output Impedance:	600 or 150 ohm balanced or unbalanced.
Frequency Resp.:	Dry channel - ±1 db 20 Hz to 20 kHz. Rev channel - from 50 Hz to 6 kHz adjustable to ±15 db.
THD:	Less than .4% maximum .1% at line levels.
S/N:	65 db.
Equiv. Input Noise:	-125 db.
Decay Time:	Selectable from 3 seconds to 5 seconds.
Controls:	On and Off switch. Decay time selector. Meter selector switch. Input gain control. Low frequency eq control. High frequency droop control. High frequency boost selector switch. High frequency boost control. Pre-mix control #1. Function selector switch. Rev mix control.
Power Requirements:	110 V AC 30 watts.
Dimensions:	Electronics - 3 1/2"H x 19"W x 10"D. Delay Line Assy. - 3 1/2"H x 19"W x 10"D.

ARCHITECTS AND ENGINEERS SPECIFICATIONS: REVERBERATION DEVICE

The reverberation device shall be completely self-contained on two rack-mountable chassis 3 1/2" high by 19" wide by 10" deep. The units shall have brushed aluminum front panels. It shall have a locking mechanism to prevent generator vibration and damage in transit.

All electronics shall be solid state and contained on two plug-in PC boards. The device shall have an input impedance of 600 ohms, balanced or unbalanced. It shall be capable of operating at an input level as low as -30 dbm and shall have an output capability of +18 dbm.

The reverberation device shall provide complete equalization of the reverberated signal, instant selection of three signal decay time constants, remote selection of three mix levels and continuous local mix control.

The device shall have a frequency response on dry channel within ±1 db from 20 Hz to 20 kHz and on reverberated channel from 50 Hz to 6 kHz, adjustable to ±15 db.

The device shall operate from a power source of 110 V AC, 30 watts.

The reverberation device shall be FAIRCHILD Model 659A REVERBERTRON.

The Electro-Voice "cocoon" of acoustically transparent material holds the mic ⅛-inch off the floor. Circle 53 on Reader Service Card.

Newest microphone from Gotham Audio is this Neumann U-47 field-effect transistor model. Circle 54 on Reader Service Card.

Compact ultradirectionality is offered in this Sennheiser MHK-815 condenser microphone. Circle 82 on Reader Service Card.

Scientific Electronic Systems' booth was filled with their special power supplies, mics, etc. Circle 89 on Reader Service Card.

This special Otari unit quality controls 8-track cartridge tape before it's loaded. Circle 61 on Reader Service Card.

Ampex showed much, among which was this film sync system from an MM-1000 to a projector. Circle 50 on Reader Service Card.

Among Pentagon's dubbing units can be found this high-speed reel to reel duplicator slave group. Circle 68 on Reader Service Card.

Teac is now providing a variety of Dolby-B encoder-decoder, including this AN-180 system. Circle 70 on Reader Service Card.

ITI has a tape loader that spools from reels directly into empty cassettes. It cuts and trims. Circle 85 on Reader Service Card.

Ferrograph stereo tape recorders are now available with built-in Dolby-B circuitry. Circle 88 on Reader Service Card.

The Scully 100. Sixteen tracks on two-inch tape with only the ancillary equipment that you want. Circle 92 on Reader Service Card.

New in the 3M tape line is 208 and 209, both designed for low print and low noise use. Circle 77 on Reader Service Card.

← Circle 24 on Reader Service Card

Electro Sound has the capability of supplying multi-track recorders to your equipment. Circle 91 on Reader Service Card.

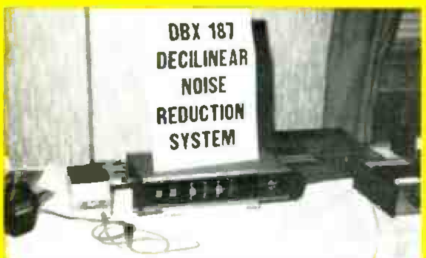
A wide variety of audio tapes for both mastering and duplicating purposes come from Agfa. Circle 58 on Reader Service Card.

Owners of Dolby 360-series units will want to have one of these full-parameter testing jigs. Circle 76 on Reader Service Card.

Tapemaker Sales specializes in making a wide variety of leader and splicing tapes. Circle 60 on Reader Service Card.

Two models of flutter meter originally made by Data Measurements are now distributed by 3M. Circle 86 on Reader Service Card.

Acoustic Research demonstrated their AR-LST monitor's six positions of frequency response. Circle 94 on Reader Service Card.



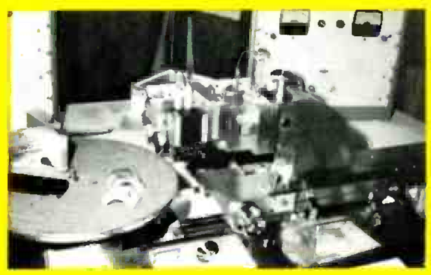
The DBX Decilinear noise reductions system joins their line of other products. Circle 95 on Reader Service Card.

Nor-Amp is a noise reduction gating amplifier for suppression of unwanted noise distributed by Harvey Radio. Circle 87 on Reader Service Card.

The Model 500 Marantz stereo power amplifier delivers 250 watts continuous per channel into 8 ohms. Circle 71 on Reader Service Card.

Crown offers the biggest amplifiers. The model M2000 will deliver 2 kilowatts into 8-ohm loads. Circle 75 on Reader Service Card.

Koss Electronics uses this kind of rig to test the response of their stereo earphones. Circle 74 on Reader Service Card.



Haeco's new model stereo cutter is shown mounted on a Scully cutting lathe. Circle 72 on Reader Service Card.

ROBERT C. EHLE

Live Electronic Music Equipment

In this article, the author explores some of the techniques and the equipment necessary to create and perform electronic music and/or electronic music plus live music in concert.

A typical? electronic music setup. This one is at the North Texas State University Electronic Music center. A lot of recognizable audio equipment is there as well as the two Moog synthesizers.

WE ARE TODAY witnessing the gradual electrification of music. Little by little, all the familiar forms of music are being taken over by electronics: organs, guitars, electric pianos, amplified instruments, synthesizers, recordings with electronically-synthesized reverb. and stereo and finally totally electronic music. Electronic music has generally been produced in studios through much effort in tape splicing and dubbing and certainly not in real time. Recently musicians, having gained a better idea of what electronics can do for them have begun experimenting with equipment on stage for real live electronic music, either alone or with more familiar instruments.

Today, the Moog synthesizers are concert instruments as are many other brands of synthesizers. Also, the rock groups, *avant-garde* improvisation ensembles, and the local "Pops" concerts have presented live electronic music in concert. Usually the instrument employed is a synthesizer; however, the synthesizer is far from an ideal instrument for live performance. Most are too slow in set-up and changing time. Most have complex functions that take many tries to get the best result which should then be spliced into a finished composition on tape. The amateur musician desiring to add electronics effects to a live performance is better advised to consider certain specific devices that will produce some reliable effects for him. This article describes several such devices.

Robert C. Ehle, Ph.D, is an authority on electronic music production. He teaches the subject at the University of North Colorado.

FREQUENCY DIVIDERS

A device on the musical market at the present time is an electronic attachment for reed instruments[1]. With a contact microphone pick-up on the mouthpiece of either a clarinet or a saxophone the player has at his fingertips (at the push of a button) the sounds of any clarinet, saxophone, double reed instrument, or cello. This device is intended for the popular music player. However, there is a basic principle at work here which serious composers can use for their own more diverse interests.

The basic principle is that of an electronic organ (most brands) and is as follows: A basic, one octave, set of oscillators is provided to generate the highest octave of notes for the instrument. Then, in all the lower octaves, instead of having more oscillators, the circuits used are frequency dividers, which take their inputs from the next higher octave. Thus, all octaves are exactly the same in intonation and the complexity of the tuning procedure is greatly reduced. One thing to note about this is that there

[1]Selmer's Maestro is one of many such devices on the market.

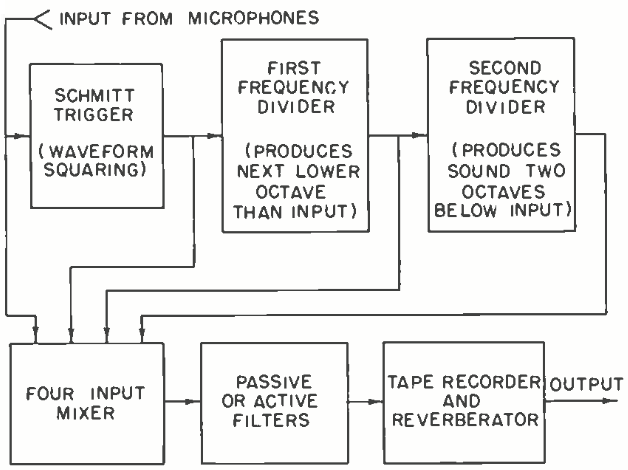


Figure 1. A block diagram of the electronic device discussed in the text.

will be no output from any octave if the original oscillator is turned off.

Now, what would happen if, instead of an oscillator, we used the output of a musical instrument with a microphone attached? We could produce any sub-octave of the basic instrumental pitch. In fact, we could have any or all octaves simultaneously. This is the principle of registration on electronic organs. These various octaves are mixed to produce various tone colors. Then, filters may be applied to the results as well as the other common signal processing devices: modulators, reverberators, mixers, and so forth. (Some of these other devices will be described later in this article.) The final result may be reproduced through a high fidelity sound system, a high-power p.a. system, or recorded on tape.

The output of the dividers may also be fed to an envelope follower or a ring modulator which is being controlled by another instrument. These will also be described subsequently.

The illustration in Figure 1 gives an explanation of how the circuit works. The schmitt trigger circuit converts the input waveform into a square wave. The two following dividers reduce the octaves one per circuit for a total of two octaves if two dividers are employed. The four signals (the input, the squared input, the first octave and the second octave) are all fed to inputs of a four input mixer so that any signal or any combination of signals may be selected.

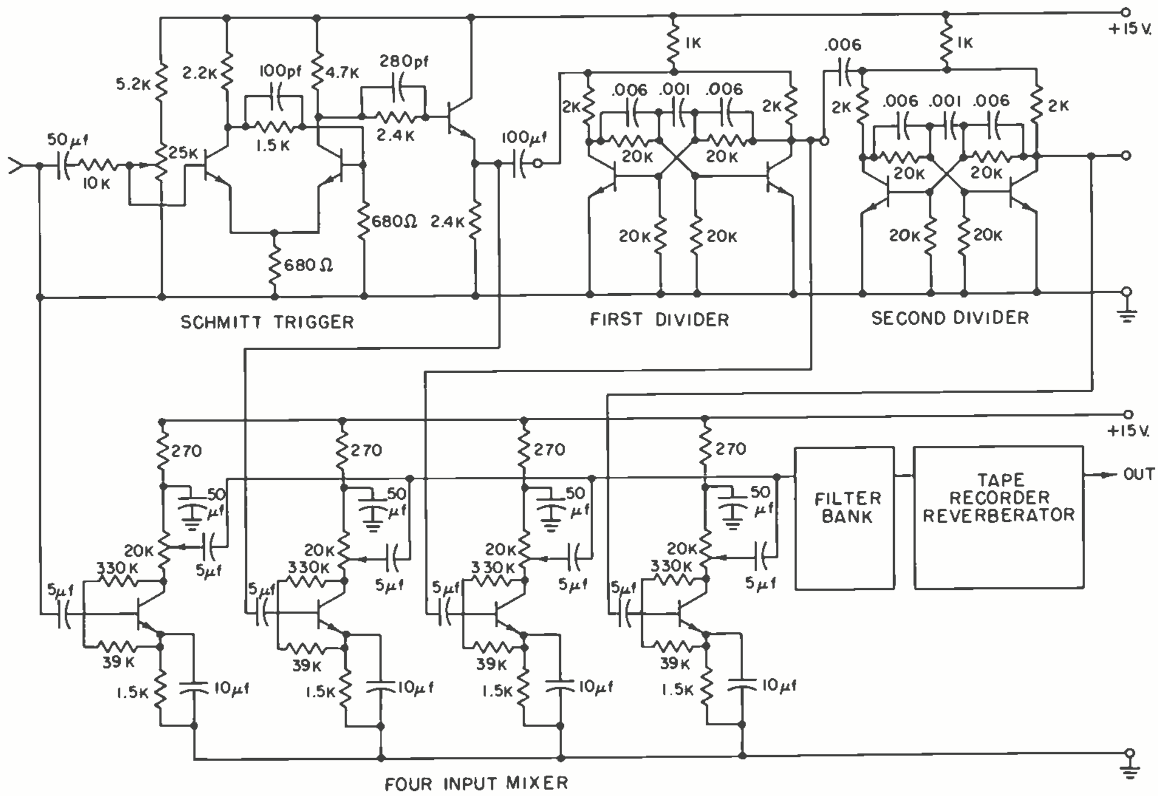
Figure 2 gives a possible circuit diagram for a frequency divider set, whereas Figure 3 illustrates how the fixed filter bank may be designed.

THE BODE-MOOG RING MODULATOR

An instrument designed by Harald Bode and manufactured by the R. A. Moog Co. is the multiplier-type ring modulator with squelch. Producing a much lower distortion output than equivalent circuits set up on the Moog synthesizer, this unit is ideal for live performance and produces a particularly attractive and musical output. The inputs may be any combination of audio signals; however, if one of the two is a live acoustical instrument, and one an electronic sound source, attractive variants on the acoustical signal result. The technique is desirable for use in real-time electronic music performances.

The multiplier-type ring modulator generates very small amounts of harmonic and intermodulation distortion (other than the desired sum and difference). Thus, when two sine waveforms are presented to the two inputs, the output is also a pair of sine waves, the sum and the difference of the two inputs. Other ring modulators may generate harmonics of the modulation products as well as products of the harmonics of the input signals. These roughen the sound

Figure 2. A possible choice of circuitry to implement the block diagram of Figure 1.



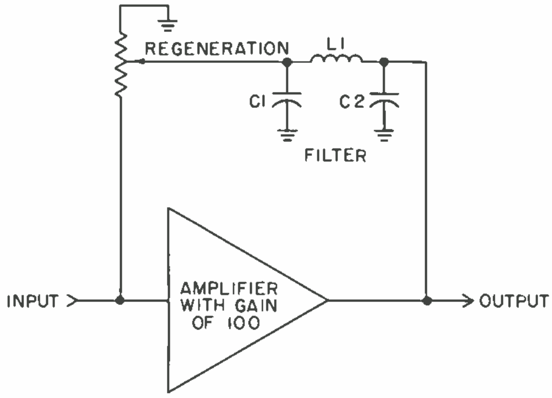


Figure 3. The use of an amplifier as a filter, regenerative filter, and oscillator.

quality of the output and make it much less attractive to the listener.

The reader is referred to the *Electronic Music Review, Volume 1, Number 1*[2] for detailed discussion of techniques that may be performed with the ring modulator. Some of these techniques will be repeated here for convenience. All of these techniques are suitable for live performance.

If one input is a band of white noise, and the other a square or sawtooth waves, having an infinite series of partials, the resultants will be two infinite series of partials, one the sum series and the other formed by the difference.

If one input is a band of white noise, and the other a sine wave, the band of white noise will appear in the output, but twice as wide, or in two segments, formed by the sums and the differences, respectively. If the sine wave is made to sweep, the white noise in the output will also sweep, producing the effects of howling wind.

Sounds picked up by a microphone may be ring-modulated with themselves if the overtones are filtered from one input and the entire spectrum is presented to the other input.

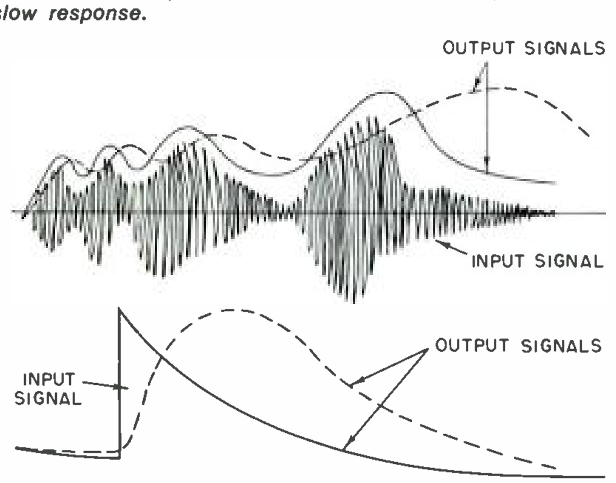
The ring modulator produces a particularly attractive tremolo effect when connected in the same manner as standard tremolo or vibrato on an amplitude modulator. If the unmodulated signal is reproduced through one loudspeaker and the modulated signal through another, the result is a sort of amplitude-phase modulation in space which is very attractive.

Any sound may be transposed (up and down simultaneously) by modulation by a sine wave. Up transposition is produced by the sum tones and down transposition by the different tones.

The Bode-Moog Ring Modulator is also available in dual units for stereo operation and for other techniques requiring two modulators.

[2]Electronic Music Revie, Volume 1, Number 1, Various authors. Discussions of applications of Ring Modulation.

Figure 4. Examples of outputs from an envelope follower with fast or slow response times. Fast response is shown with a solid line, while the dashed line indicates slow response.



ENVELOPE FOLLOWER

An envelope follower is an adaptation of a device common in radio applications known as a detector. Its function is simply to detect the lower of two modulated signals. That is, to separate the lower of two signals from the modulation product after modulation has been performed. Although the lower signal is usually selected, a detector for the upper signal could be designed and built. A rule to remember in using detectors or envelope followers is that the signals to be separated must be several octaves apart. This is because the actual separation is done by filters, and filters have a limited selectivity (half-octave or more, usually).

The envelope follower, while identical in circuitry to a detector, is usually employed to separate more complex signals, that is, signals with more than two component elements. These signals are often complex audio signals containing hundreds of component frequencies. Such signals are not the resultants of a modulation process, however but are treated by the envelope follower as if they were.

In actual practice, the envelope follower is used to extract a low-frequency signal from a complex audio signal. This low-frequency signal follows the general outline of the complex audio signal with a greater or lesser degree of accuracy depending on the settings of certain filters which are usually made variable and which may be labeled *slow*, *fast*, etc.

The envelope follower output signal is employed as a control voltage on voltage-controlled modules, thereby imparting the envelope extracted to other audio signals. This is illustrated in FIGURE 4.

TAPE RECORDERS AND REVERBERATION DEVICES

Once the experimenter has assembled a group of sound generating and modifying components (such as those previously described, or a professional synthesizer) he will probably find that his biggest problem is the obtaining of a suitable recording system. No matter what convenient or inexpensive means are used to generate the sounds, they are usually recorded for presentation as sound tracks, phonograph records, and so forth. Thus, tape recorders are required, and for professional applications a professional tape recorder will be needed. These cost much money and can not be improvised inexpensively. Most professionals recommend that the electronic music studio have at least two recorders and one of these should have four or more channels. For live electronic music one machine may be used as a reverberation unit and another for recording the performances.

The amateur should be able to purchase two good tape decks and build recording and playback amplifiers to use with them. He can purchase four-track heads for either quarter-inch or half-inch tape and the head manufacturers will usually provide circuits for amplifiers. This is the only economy move recommended since tape decks are notoriously difficult to design and less than high quality here shows up in poor speed regulation and other problems.

The diagrams of FIGURES 5 and 6 illustrate the head configuration of a professional tape recorder with *sel-sync* (an Ampex trade name) and a method for adding an extra

Recording head Playback head	Impedance	Voltage	Current
Recording head	high	high	low
Playback head	low	low	high

Characteristics of tape recorder heads in professional recorders.

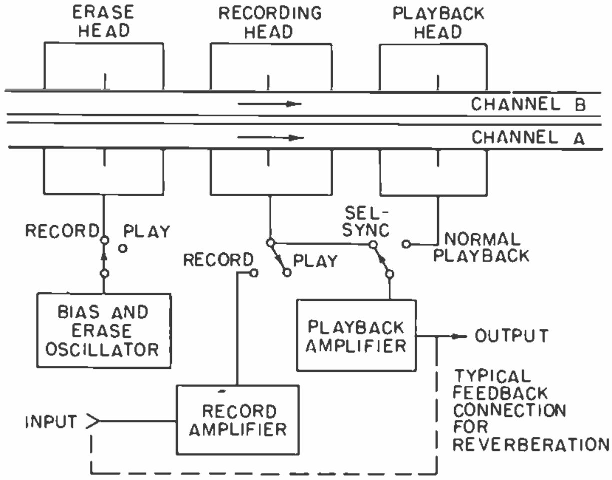


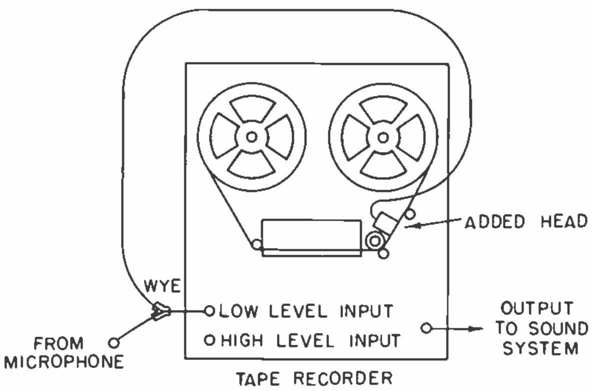
Figure 5. A typical tape recorder with sel-sync and three heads as used in professional electronic music studios.

head to a two-head machine to create tape-recorder reverberation. The illustration of the steel spring in FIGURE 7 shows another popular reverberation device that may be used.

The studio with sufficient funds is advised to invest as much as possible in high-quality tape recording equipment. An expensive but useful addition to the professional tape recorder is the Eltro Pitch and Tempo Changer which can be adapted to most professional machines. This device (FIGURE 8) has a rotating head assembly containing four heads and a variable speed capstan. It can be used to change both pitch and tempo or either one independently. A less expensive and less versatile device is the variable-speed tape capstan drive oscillator-amplifier manufactured by various companies, including the R. A. Moog Co. With this device, the pitch and tempo are continuously variable together but not independently.

The technique of mixing the two signals from two recorders, one with variable speed and one without may be used to produce the phase-shift effect popular in many current rock recordings. The cancellation frequency is shifted by changing the signal delay in the recorder with variable speed and beating it against the unchanged signal in a simple mixer. If both recorders are running at exactly the same speed there will be no phase difference and no

Figure 6. The method of adding a head to a tape recorded to use for reverberation.



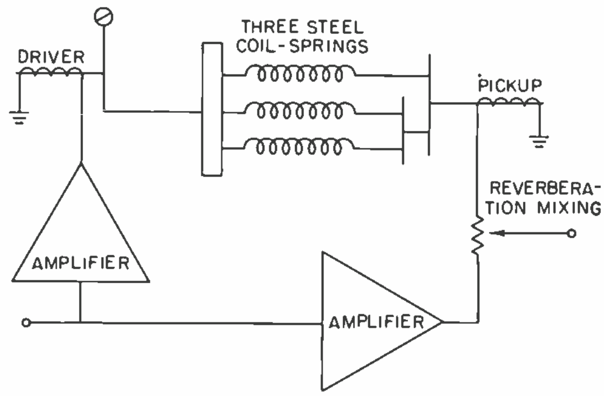


Figure 7. A typical steel spring reverberation device.

phase cancellation effect. This technique causes difficulties in live performance, however, due to the inherent time delay involved. A similar but not delayed effect may be achieved with either a ring modulator or a resonant voltage-controlled band-pass filter (both have been described earlier in this article).

MONITOR SYSTEMS

Two types of amplifier-speaker systems must be considered for electronic music playback. The first are smaller units for studio use. These may be regular high fidelity units. Vacuum-tube versions are recommended because they may often be obtained inexpensively today and also because they will stand up to experimental and accidental misuse better than transistor amplifiers.

The second type of monitor system to be considered is the large, high-volume type necessary to produce the volume levels for filling large auditoriums and theaters. Amplifiers for such a system are best transistorized for portability and should produce at least 50-watts per channel. Acoustic suspension speakers are often used where portability is an important factor but these are low-efficiency systems and much more sound can be obtained from large horn systems. Popular music groups often use heavy-cone, fifteen-inch speakers in open-back cabinets. These produce a great deal of acoustical power but add an undesirable coloring and distortion to the sound for some applications.

TAPED ELECTRONIC MUSIC

Taped electronic music is most popular today on the college and university campuses and is the primary method used by serious electronic music composers. It should be obvious that it is possible to achieve a great complexity by means of splicing small segments of tape together. If each segment contains a complex sound and segments containing diverse sounds are spliced together in fragments of less than an inch long per sound, the result is almost unbelievably complex. With some practice it is possible to anticipate the result of mixing pieces of tape from various recordings in various fashions. For example, take a tape of a recording of a high-frequency sound which is very complex (such as the sound of a cymbal crash). Cut this tape into one-inch segments and splice these between one-inch segments of a low-frequency recorded on another tape. The result will be several times as complex and interesting as either sound originally was. Then try a similar experiment but start with five-inch segments and gradually reduce the length of the segments until they are a half-inch or less in length. This experiment can be varied by using changing tones rather than constant ones. Also the finished tape can be dubbed (copied) and then the copy can be cut up and interspersed with another tape.

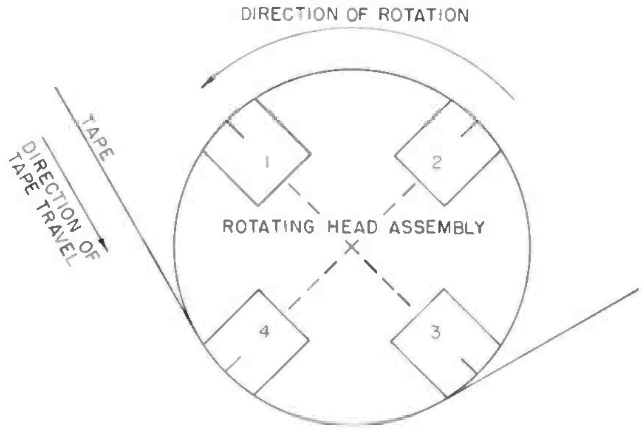


Figure 8. A tape player with a four-head rotating group used for independent changing of pitch and tempo of a recording. When the head assembly rotates clockwise, frequencies on the tape are increased as small segments are repeated; rotated counterclockwise causes frequencies to reduce and small segments are skipped.

Of course, the best results will be obtained in this sort of process if the person assembling the tape has specific goals in mind. For example, start with simple sounds and gradually build up to a climax in the musical composition by gradually increasing the complexity of the sounds, the shortness of the tape segments and the contrast between the materials on the tapes being spliced. Remember that you can change the speed of any sounds on your tapes by copying them at different speeds than they were recorded. Tapes can also be played backwards (particularly interesting for musical instruments and voices) and spliced together with alternating forward and backward sections.

Many techniques for splicing have been developed. For example, splice a long gradual splice by cutting a long diagonal cut in two pieces of tape and splice the two sections together with splicing tape. If one uses splices like this often it pays to construct a special template that will fit a professional splicer and make cutting straight edges easier. This can be a strip of metal the width of the tape with a cutting edge of the shape of the desired tape splice. Two pieces of tape cut with the template will match exactly when spliced.

Remember that it is possible to splice sections of blank tape in between sections of recorded tape and that these blank sections can be as short as a half inch for short interruptions within a continuous sound. This can produce interesting rhythmic effects.

As was mentioned earlier, for best results do your work at fifteen inches-per-second on half-inch tape with four channels. Still, good results can be obtained with home tape recorders; even if they are not up to broadcast standards they can be very satisfactory for amateur performances, background music for local theater groups, sound track for semi-professional movies and also home movies, local dance groups and so forth. They can also be quite adequate to provide a great deal of satisfaction and entertainment.

Taped electronic music is not live, but can often be mixed with live music to produce live results. For example, it may be used with an orchestra or as the background to a ballet or the theater. ■

STEPHEN H. LAMPEN

Four Channel on the Road

Do you really need to cart sixteen tracks along to record quad on a remote. The author says no, and makes a good case for taking no more than four channels along.

THREE LITTLE WORDS send most hi-fi enthusiasts reaching for their pocket books or preparing to remove their whole system. Others, who have been content with their tube-type mono systems for twenty years, wave these three words off as yet another fad. The three words are four-channel stereo.

It's been called surround sound, quadraphonic, quadrasonic and (by those still content with one channel) four times as much noise. Whatever one may call it, four-channel sound seems to be here to stay. To the uninitiated, a four-channel recording session summons up visions of huge, multi-channel consoles, gigantic sixteen- or twenty-four channel recorders, a gross of microphones and engineers with twenty-seven fingers. These visions, however, seem to have their basis in fact as most large studios are using the multi-channel recording and mixdown for quadraphonic recording. This need not be the only solution. A straight, four-channel recording contains only five essential parts: A four-channel recorder, four microphones, a four-channel mixer, a ten-fingered engineer, something to record.

With these five criteria in mind, two San Francisco-based companies have joined forces on numerous occasions to produce four-channel recordings. 3P Recording supplied the recorder while Hansonic supplied all the mixers, microphones, cables and monitoring equipment.

Remember the days when mono recordings were produced on mono recorders? It was not so long ago that stereo recordings were made with a studio's two-channel Ampex or other workhorse. Does it not make sense, therefore, to carry these examples to their logical conclusion: four-channel recordings should be made on four-channel recorders? This does not mean that multi-channel recordings are a waste of time; the reasons for multi-channel recordings are well taken: increased versatility with the subject matter, greater working area on tape and complete control during mixdown. But, in many instances, these advantages can be traded for even greater advantages in doing a straight four-channel recording.

3P Recording specializes in on-location mastering and, in many instances, the advantages of a straight four-channel recording are even more evident. The first, and most obvious, advantage is weight. The four-channel recorder, a Vega 800-4, shown in FIGURE 1, is a virtual copy of the Ampex 440-4. It is the largest recorder available in portable cases. After carrying it for any appreciable distance, the reason become quite apparent: the electronics and the deck weigh a total of one-hundred and fifty pounds. This makes one hesitate to use the word *portable* when describing the recorder. Needless to say, the eight or sixteen trackers do not come in portable cases, making their exteriors more vulnerable to bumps and scratches. These machines cannot be carried from place to place but must be carted or rolled. Most of them are too bulky to fit with a multi-channel console in any but the largest rooms when on-location. The size of recorders is continually decreasing and multi-channel machines can now fit into smaller places than before. But even these weigh a quarter of a ton and hand-carrying them is out of the question without a small army.

The second advantage is tape. Most four-channel recorders use half-inch tape, as does the Vega 800-4. Though expensive, a half-inch reel nowhere compares with the cost of one or two-inch reels. Also, the weight of a two-inch reel compared to a half-inch reel is considerable, but only if someone tries to carry a dozen reels at once. Still, even in the largest sessions, some junior engineer will try to move all the tape in one go—and boxes go flying (or, rather, falling).

Stephen H. Lampen is the owner of 3P Recording of San Francisco, California.

Figure 1. The Vega 800-4 in its portable two cases.

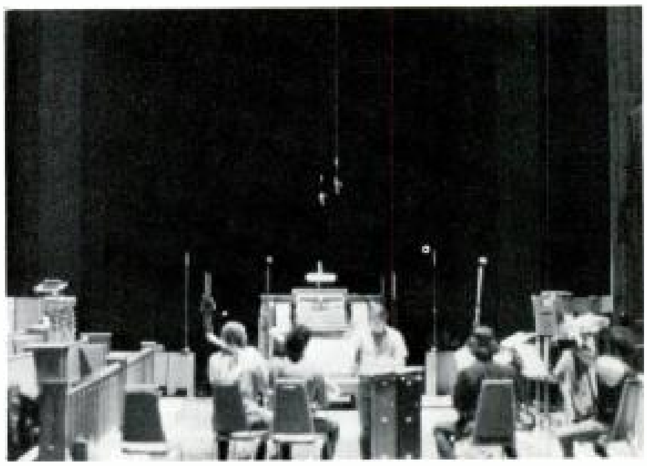


Figure 2. The mic placement described by the author. The two hanging mics are part of the cathedral's sound system, not part of the recording.

The third advantage, which large studios on-location rarely take into consideration, is the source of power. A large session may draw as much as fifty amps. This could be too much for a building with 1920-style wiring and either fuses are blown or necessary equipment is turned off for lack of power. On the other hand, a four-channel recorder with a small console (or a pair of mixers) and a monitor system rarely draws more than five amps. This means it can be set up in small, out-of-the way places with just one a.c. receptacle.

The final advantage is the cost of the equipment itself. A $4000 four-channel recorder is $10,000 to $20,000 less than its bigger brothers. A small four-channel console is as much as $30,000 less than the gigantic multi-channel boards being used. And two good stereo mixers can reduce the price even further.

What, then, are all the advantages to a company which really does have an army of engineers and which cares not about the cost of a reel of tape or a recorder and which never encounters power problems? The answer is: in many instances, four channels are all that is needed and anymore channels are redundant. To illustrate this, the set-up for a four-channel session will be used as an example.

Many of the recordings made by 3P Recording and Hansonic are at Grace Cathedral in San Francisco. This huge Episcopal church, atop Nob Hill, has inside dimensions which approach infinity! The church is almost 300 feet long and the ceiling measures just over 92 feet in height. The decay time approaches seven seconds! This produces an acoustical environment both unique and challenging. The decay time is unsuitable for many types of music, especially rock. Even the large organ, at times, is overcome by the excessive amount of acoustical reflections in the building. It is essential, in a building such as this, that the acoustical effect of the building be included in the recording. There are two simple reasons for this: first, a different quality of sound may be one of the primary criteria for going on-location and, second, with a building like Grace Cathedral, it is impossible to avoid getting large amounts of reverberation on tape, so it is necessary to get it on tape in the best way possible.

In the example set-up the *Royal Court Brass*, a classical brass group, was the subject matter for a series of quadraphonic recordings. A tight brass sound sounds excellent with a large amount of reverberation. FIGURE 2 shows the microphone placement. Similar to the Blumlein method of co-incident microphones for stereo recording, two mics, Electro-voice RE-20's, were placed on Atlas MS-25 stands with BB-1 booms. These mics were our front two channels and were placed about ten feet in front of the group. In the background, also on MS-25 stands with booms, were a pair of Telefunken U-67 condenser mics. These were placed on existing granite partitions to increase their height. They were placed about thirty feet behind the front mics; close enough that there was little lag between front and rear mics but far enough away that these mics picked up predominantly the natural reverberation (ambience) of the church. Needless to say, these were our back two channels. The performers were set up in a semi-circle to increase reflections so that the back mics would pick up less of a direct image. It is interesting to note that, with just four microphones, it takes less time to set them up and move them around until the correct balance is achieved than it would if one had to set up one mic for each instrument and a couple for the ambience. The hanging mics in FIGURE 2 are Sony C-37A's and are part of the church sound system; they were not used in the recording.

FIGURE 3 shows the mixers as they were used for this recording. An Ampex mixer and two Shure M-67 mixers were used. The Ampex mixer, with its 600-ohm balanced line outputs, is a direct match to the Vega and most other large machines. The Shure mixers also have 600 ohm outputs but they are not Cannon connectors and require adaptor cables. As the total cost of these mixers is slightly over $500, this represents a significant saving over a small console while the quality of signal is still acceptable. As an added advantage, the Shure mixers have internal oscillators with which levels can be set on both the mixers and the recorder.

All the recording equipment was set up in a room right next to the chancel, where the group was playing. Separated by a hall, this was an ideal spot as no cable ran more than a hundred feet and the engineer was close enough for easy contact with the performers without adding the sounds of the recorder in operation to the tape itself. All the cables used were Belden 8412 or 8428, which has thicker insulation, with Cannon XL plugs. Having tried 8414, which has a poorer shield, the risk of interference (especialy on-location where mic lines are often tangled up with a.c. cords) was not worth the difference in price.

FIGURE 4 shows the monitoring solution: headphones. Dick Garretson, owner of Hansonic, is shown mixing a selection while monitoring levels on Superex headphones. These are 600-ohm headphones as the auxiliary outputs on each of the electronics is 600-ohm out. Ampex amplified speaker systems have also been tried with excellent results. However, headphones presented an interesting problem. Since they can normally monitor only two channels, these are usually the front two channels, which contained the actual program material. As they were not hearing the back microphones, the end of a selection was often reached and the *stop* button pushed before the reverberation had fully fallen. After the last note is sounded, a wait of almost four seconds is required. The simple solution might be to monitor one front channel and one back channel. But this produced confusion to the engineer, since the balance between the channels on the phones was very disturbing. The only solution left (aside from buying four-channel headphones) was to patch the front and rear on each side together by using adaptor cables. This has proved to be very successful.

Needless to say, this is not the only kind of recording being produced today. How could one use this basic idea for rock or other popular music? 3P and Hansonic have done a number of rock recordings using a version of our original criteria: four-channel recording on a four-channel recorder. In these cases, the basic system can still be used effectively without having to resort to more channels—and more trouble. Most four-channel recorders offer sel-sync or overdubbing like their bigger brothers. The ability to add or erase selective tracks with this is a great help.

The basic difference between a rock session and a classical session is instrument placement. In most classical sessions there is a set pattern in instrument placement and, thus, a set pattern in recording. But popular music, having no set instrument positions, offers the engineer the ability to actually structure the acoustical properties of his final product. While leakage between microphones may be present in large degrees during on-location classical recording, and may even be helpful in giving the instruments well-defined spatial locations, it is rarely desirable to have large amounts of leakage when recording rock. Leakage between mics picking up amplified instruments, and between amplifiers and singers mics, is often excessive. Leakage is especially noticeable when recording in small enclosures or in a highly reverberant room.

Figure 3. Two Shure M-67's and an Ampex mixer comprise the transportable console.

One of the solutions to excessive leakage is to spread the instruments farther apart. However, this is impossible with limited space (such as a stage) and may even lead to a poor performance as the musicians cannot act as an integral unit. Another solution is to provide acoustic dividers between instruments. To carry these when on location is very difficult, if not impossible, because of their bulk. Also, acoustic dividers are not only ugly but can fall over or be tripped over since they divide the performers. This is especially true during a live performance.

Another solution to this problem is the ability of *the recorder* to add channels by means of overdubbing. The-

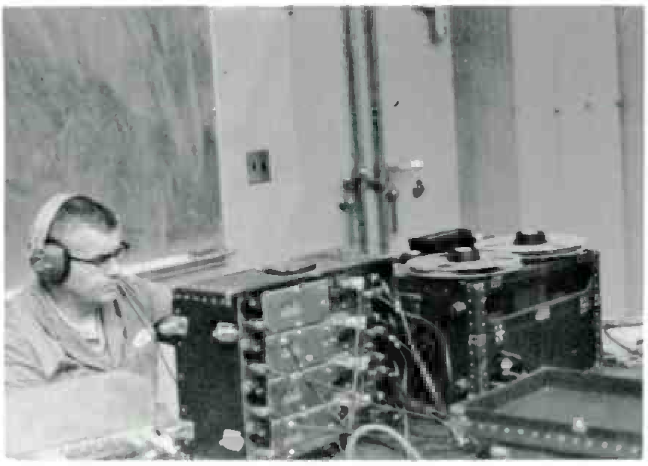


Figure 4. Dick Garretson, Hansonic owner monitors on headphones.

oretically, this could reduce the number of mics to one, simply switching it from channel to channel until they are all filled up. Naturally, this approach bids good-by to the four-mic ambience method. Multi-mic sessions being the only solution, many studios will use up to thirty mics on a large rock session. However, this seems excessive both in cost and difficulty. Ten mics should suffice if overdubbing is used.

On one rock session, 3P and Hansonic miked the two guitars and the drums on two tracks. In this case, the group had a large acoustic divider which produced excellent separation between the drums and the bass guitar. It seems, the more recordings which are attempted, that the bass guitar is the biggest offender in leakage to other mics. It is obvious then why many studio recordings take the bass direct to console. Unfortunately, this is rarely possible on-location. After the guitars and drums are recorded on channels one and two, the vocals are dubbed on channel three. Channel four was back up instruments which would have been lost in the original mix on channels one and two, such as acoustic guitar or piano. This solution to the problem of leakage works excellently and the mixdown engineer has everything under his control.

During mixdown, the engineer should be aware of one interesting advantage in four-channel recording. In stereo, there are actually three channels: left, right, and center. A-Bing a mic on the original mix will turn up on the center channel. Thus, if there is a phantom channel between every two channels on a four channel tape, there are six more acoustic channels besides the original four. Thus the mixdown engineer has ten distinct positions as he mixes down. Of course, gradation of left and right can be created using pan pots, giving the engineer an infinite number of channels with which to work.

Mix down in this case is a misnomer. Because, if this is a quadraphonic recording, one is mixing from four to four. But the method of mixing a master as described leaves the engineer a lot to do. Any panning between channels is done during the mixdown—not during the recording. It is good that anything that has been forgotten during the recording of the master can, in many cases, be added later. However, anything added by mistake is more difficult to remove. Thus, with four distinct channels (such as guitars, drums, vocals, and instrument accompaniment) the mixdown engineer does indeed have the choice as to what goes where—and how much! Mixing down is also the time to add any equalization. The only exception which can be made is to put a compressor-limiter on the voice channel during the master recording. However, this is one more thing to carry and one more thing to go wrong during the session. ■

db visits Koss Corporation

Figure 1. An outside-evening view of the Koss plant.

Figure 2. A view of the 20,000 square-foot production department.

Figure 3. John Whittaker explains the master test station used in checking the electrostatic phones.

Figure 4. Production of the Pro4 AA elements. Each element is completely manufactured by Koss using specialized machinery developed for this purpose.

Koss prides itself as being the largest manufacturer of stereophones in the U.S. and recently invited **db** to view its new plant in Milwaukee, Wisconsin. The new plant includes administrative offices, manufacturing, research and development, and shipping in a single level 60,000 square-foot building. An additional 5,000 square-foot building adjacent to the main factory houses consumer communications.

The new headquarters substantially increases Koss facilities which were formerly in a 25,000 square-foot building in another part of Milwaukee.

The new building has cantilever construction which permits flexible interiors uninterrupted by structural posts. It provides sufficient space for Koss to manufacture more than 50 per cent of the component parts which make up their line of stereophones. The center of activity is the more than 20,000 square-foot final assembly area where five conveyorized assembly lines are in operation. Adjacent to this area is a "clean room" which houses the critical manufacture and assembly of driver units. A number of testing stations perform the extensive electronic and subjective tests before packaging. At the master test station for electrostatic stereophones, nine electronic inspections plus listening tests are conducted. A machine run response curve of each electrostatic's performance is packaged with each unit as proof of performance.

In addition to an extensive industrial engineering department, Koss has a research and development department that features an anechoic chamber plus a host of sophisticated equipment for the testing of electro-acoustic products.

A Customer Communications Center is located in an adjacent 5,000 square-foot building and includes advertising, printing, customer service and package design. It also produces all point of purchase materials for some 3,500 dealers in the United States as well as all catalogs and other promotional material.

Figure 5. John Koss (right) enjoys showing a group of foreign representatives the new facility. The group is reviewing features of the new Red Devil model KRD 711.

In addition to its full line of dynamic stereophones, Koss is well known for its electrostatic phones and has recently started to deliver its model K2+2 Quadraphone, a discrete four-channel headphone. A number of headphone accessories are also produced, such as multiple listening stations, adapters, extension cables, etc.

The Koss plant is certainly an amazing growth story. From its humble beginnings in 1958 it has grown to become the largest and probably the best-known manufacturer of stereophones. It has brought the world of headphone listening to the status of more than just a component accessory for the hi-fi consumer, and to a necessary product for the audio professional. ■

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(continued from page 14)

well to see what the resolution of the system is in the corners. (Note that specifications for cameras, etc., will usually rate the system quality by resolution, and most times with the additional phrase "at the center".) The chart also has resolution lines at the corners, usually in circles. The readings at the corners will generally be about 100 lines less than at the center. This is considered normal. Therefore, if a specification for a piece of t.v. equipment just lists a resolution figure without any further qualification, that will be the center quality and the resolution at the corners can be judged to be about 100 lines less.

While viewing the chart, several other system qualities can be seen. The squares of vertical lines are also marked with numbers indicating the resolution. These squares of equal resolution should be equal on all sides in the image. If they are not of the same width and height, the linearity should be adjusted until the squares appear as square. With the linearity thus adjusted, the circles in the center and in the corners should now appear true-round.

Around the center of the chart, there are heavy (wide) black lines which change distinctly in tone. These gray scales should show ten definite blocks in all directions. Also, somewhere at or near the center of the chart, there is a very heavy black line which is constant in tone throughout its length. The edges of the bar should be clean with no after-trails. Adjustment of the peaking control will eliminate any black or white streaks. There are also four diagonal lines at the center of the pattern which radiate from the center toward the corners of the chart. These indicate functioning of the interlace system. The systems using internal or external EIA sync. generators will (or should have) straight diagonal lines. Otherwise, the system sync. signal needs adjustment. In random interlace systems, the lines will probably appear jagged, which is normal for that situation.

Television resolution does not relate to any of the other uses of the term. In optics, for example, the resolution is measured in lines per millimeter. Since television resolution relates to picture height, *not picture size* (or monitor size, either), there is no direct relationship between the term resolution in television and the definition in optics. Thus, it only seems that the dictionary definition can be applied to television, but actually the distinct reference in television of resolution to picture height (not size) makes the difference singular and unique. ■

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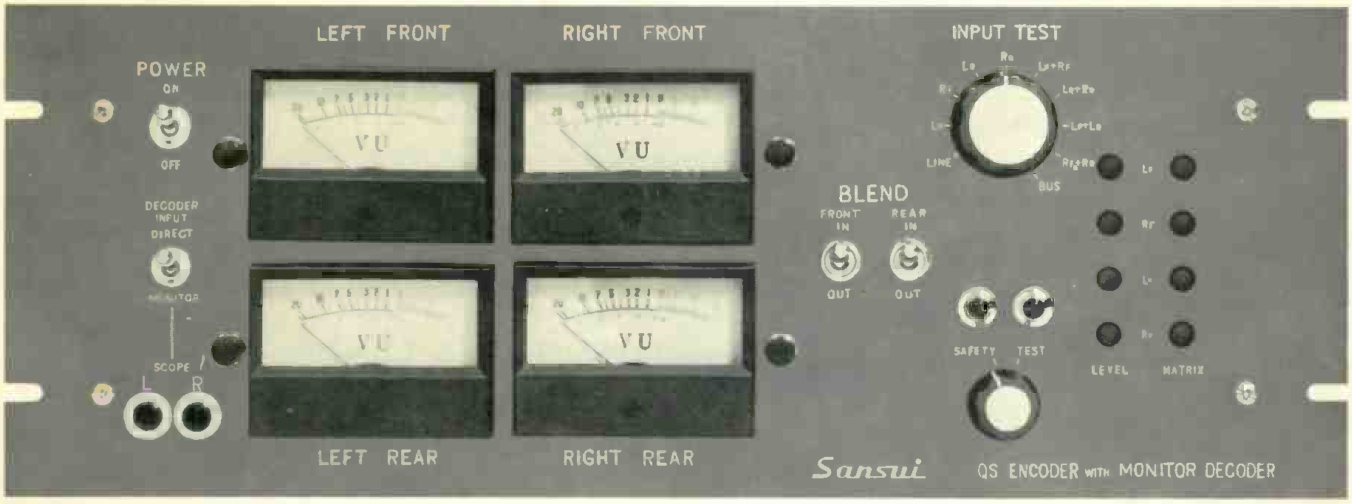

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