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THE SOUND ENGINEERING MAGAZINE

NOVEMBER 1976 $1.00

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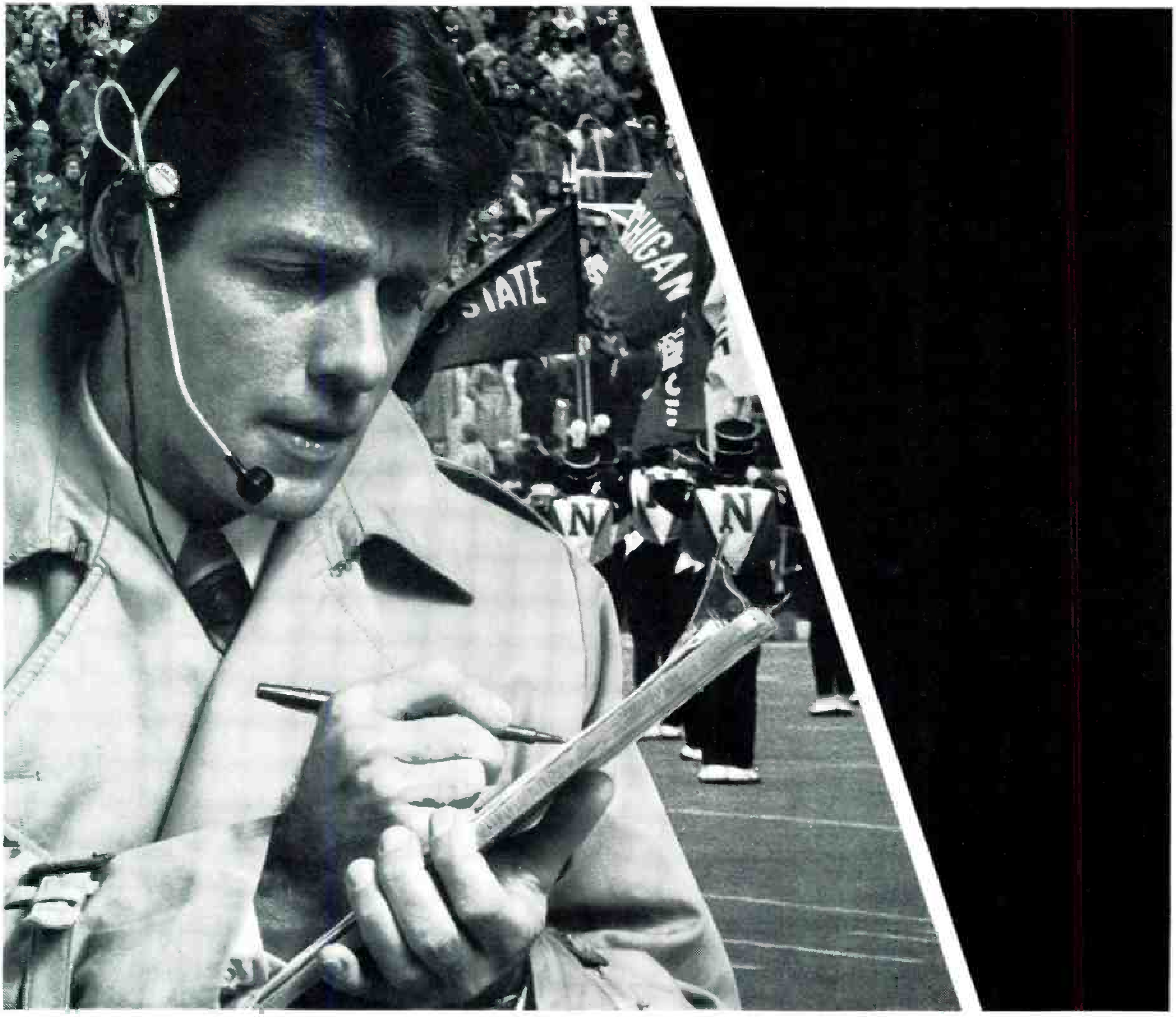

coming next month

• Space this month did not permit the inclusion of the previously promised article on the AMPEX ATR-100 SYSTEM. This detailed look at its creation and working will be in December.

• Mortimer Goldberg is technical supervisor with CBS Radio in New York City. He has been responsible during his 25-year career for the editing of many documentaries and he brings this expertise to bear in his article THE ART OF TAPE EDITING.

about the cover

• A bird's eye view of music in on-stage production.

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THE SOUND ENGINEERING MAGAZINE

NOVEMBER 1976, VOLUME 10, NUMBER 11

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

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db letters

THE EDITOR:

In my article, "Digital/Analog Level Control," in **db**'s September issue, the address zip code for Analog Devices is incorrect. The correct address of the firm is: Analog Devices. P.O. Box 280, Norwood, Mass. 02062.

WALTER JUNG

THE EDITOR:

In Robert Faulkner's article, "A Neat Little Dual Limiter," in the August issue. there appears to be a printing error in Figure 1. The 2N3053 has no bias to turn it on. Also, I wonder if Mr. Faulkner has given any thought to making the circuit into a compressor by changing the type of light-dependent resistors for a different attack and release time and the gain of the 377 for differing slopes.

HENRY L. BROOKS
Los Gatos, Ca.

Mr. Faulkner replies:

I had not considered making the attack and release time variable. My thinking was that there are too many knobs in this industry already. However, you may have a point.

I guess I went through every light-dependent resistor known to man and decided on the 704L-2 because it had a very fast attack time and a reasonable release time—three seconds for 75 per cent recovery. Also, the 704L-2 had the most linear response, which is important in this configuration.

The 2N3053 does not have a bias resistor to turn it on because it is not used as an amplifier. It becomes a variable resistor the way it is connected and therefore controls the brightness of the led.

ROBERT R. FAULKNER

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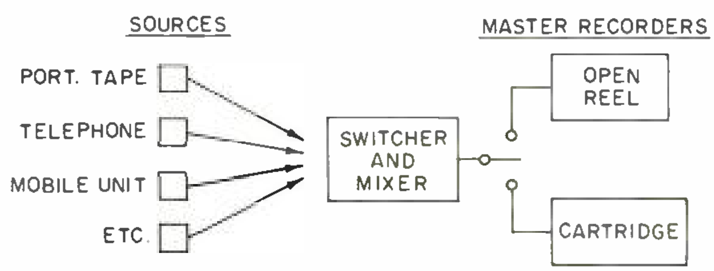

Fig. 1. When a variety of sources must be channeled to one or more master recorders, some type of switching and mixing unit is needed.

• Many electronic means are used for gathering and editing the news presented by radio stations. Most of the effort ends up on audio tape for air presentation over the station's regular facilities. The news room is actually a specialized, but major, recording facility of the station. Although the bulk of news programming is talk in nature, the technical quality of the finished product should be comparable to that offered by the station's other facilities.

BASIC PROBLEMS

When a variety of sources must be channeled into a master recorder, some means of switching and mixing is desirable. A standard console can be used for the purpose, or (as is often the case), the station can design its own arrangement. Whatever method is used, impedance matching of the various equipment units, and signal levels throughout the system must be considered.

The input/output impedances of each unit are some of the basic design parameters which allow various units to electrically interface. Impedances

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should be matched properly so the most efficient transfer of the signal from one unit to another can be accomplished. When units are connected together without consideration for impedance matching, then a serious mismatch can occur, affecting both signal levels and audio frequency response.

Signal amplitudes are important both from an operational as well as a technical standpoint. In operation, if the levels are at great variance with each other, the operator will have a difficult time editing and recording master tapes. If levels are too high, amplifiers may be overloaded and distortion results. When levels are too low, then there is the chance of cross-talk, noise, and interference problems.

HEADROOM

Signal levels and distortion often go hand in glove together. One way that distortion can occur is through lack of *headroom* in the equipment design. True signal peaks are not indicated on the vu meter; they can be 8 to 10 dB higher than the indicated peaks. Headroom is that region above the indicated signal peaks and the point the amplifier goes into distortion.

You can check out the headroom in the equipment or system with a sine wave audio signal and a distortion analyzer. Feed the tone into the input of the system and adjust controls for normal operating levels. Measure distortion at the output of the system. Without changing any of the adjustments, increase the output of the signal generator by 10 dB and again measure distortion. If there is no change in the measured distortion, then there is adequate headroom in the equipment. But if the distortion increases significantly, headroom is lacking. In this case, set normal operating levels *downward* to a new value for normal operating levels. This should be at least 10 dB below the point where the amplifier or system goes into distortion. One word of caution: either pad the vu meter or disconnect it during the high level test to prevent damage.

PADS AND TRANSFORMERS

Match impedances and control signal levels throughout the system by use of resistor pads or transformers. Precision pads may be purchased, but home made pads are usually adequate. The resistor values for the pads may be calculated from formulas found in many technical books or, oftentimes, equipment manufacturer catalogs will provide tables of resistor values for building pads. These are calculated values and may not always match up with standard stock resistor values. So, select stock values close to the required values. This will affect the end results by a few dB, but otherwise they will be satisfactory.

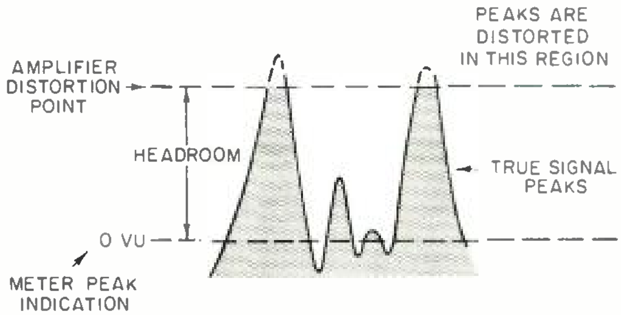

Figure 2. Headroom is the space above the signal peaks, as indicated on the vu meter and the point the amplifier begins to distort.

Transformers can be used for isolation, matching, or bridging. Good quality audio transformers are somewhat expensive, so you may desire to stick with resistors to do the job. But remember that resistors always create a loss when placed in the circuit, so the equipment should have enough reserve capacity to make up for this loss. When isolation is a definite need, it's best to use a transformer.

HEAD ALIGNMENT

Recorder heads, especially those on the master recorders, should be kept in proper alignment. Use a standard NAB alignment tape for the playback alignment first. Then align the recorder with audio tones from a signal generator using a good tape to record the results. It is not enough to obtain good results on the master machine alone. Take the tape you just recorded with tones to one of the control room machines and play it back. The results produced should be very similar to

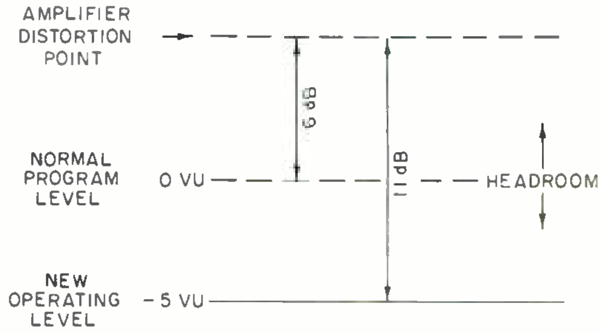

Figure 3. If the amplifier or system does not have adequate headroom at the prescribed normal program levels, then set a new operating level at least 10 dB below the amplifier distortion point.

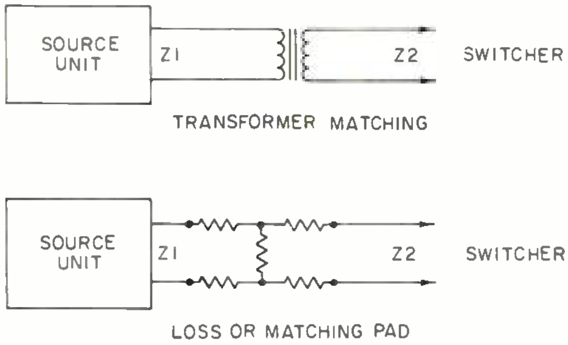

Figure 4. Impedances should be matched and signal levels controlled with either pads or transformers.

those obtained on the master machine by itself. If there is a great variance, go back and realign the master machine again. Make sure the playback equalizer in the master machine is not overcompensating for poor head alignment. Unless the recordings are compatible, the quality will suffer when the control room machines play the recorded product on the air.

CLEANING

All the news tape machines, especially the master recorders, should be cleaned on a regular basis. Clean the heads, pinch rollers, drive shaft and guides. A main problem can be caused by oxide from the tape that builds up on the head, forming ridges that prevent the tape from making tight contact with the face of the head. It is not possible to get a good recording unless there is a tight head-to-tape contact. The results will be low signal levels and poor audio frequency response.

Another problem can be iron oxide from the tape, clogging the gap between the pole pieces of the head. This will produce a magnetic short circuit to the flux lines of the head, making an easy deflected path for the flux instead of it going through the coating on the tape. (The gap is air or other non-magnetic insulation which has a high resistance to the flux lines. As the tape passes against the face of the head in tight contact, the iron oxide on the tape creates a low resistance path so the flux lines flow into the tape and thus magnetize the iron particles—the desired result of the recording.)

Also, oxide coating can harden on the head, guides and other spots where there is friction with the tape, producing a very abrasive surface that will scratch and wear out the tape quickly. If small sections of loose tape wrap around the pinch roller or capstan drive shaft, this will increase their diameters and change the tape speed.

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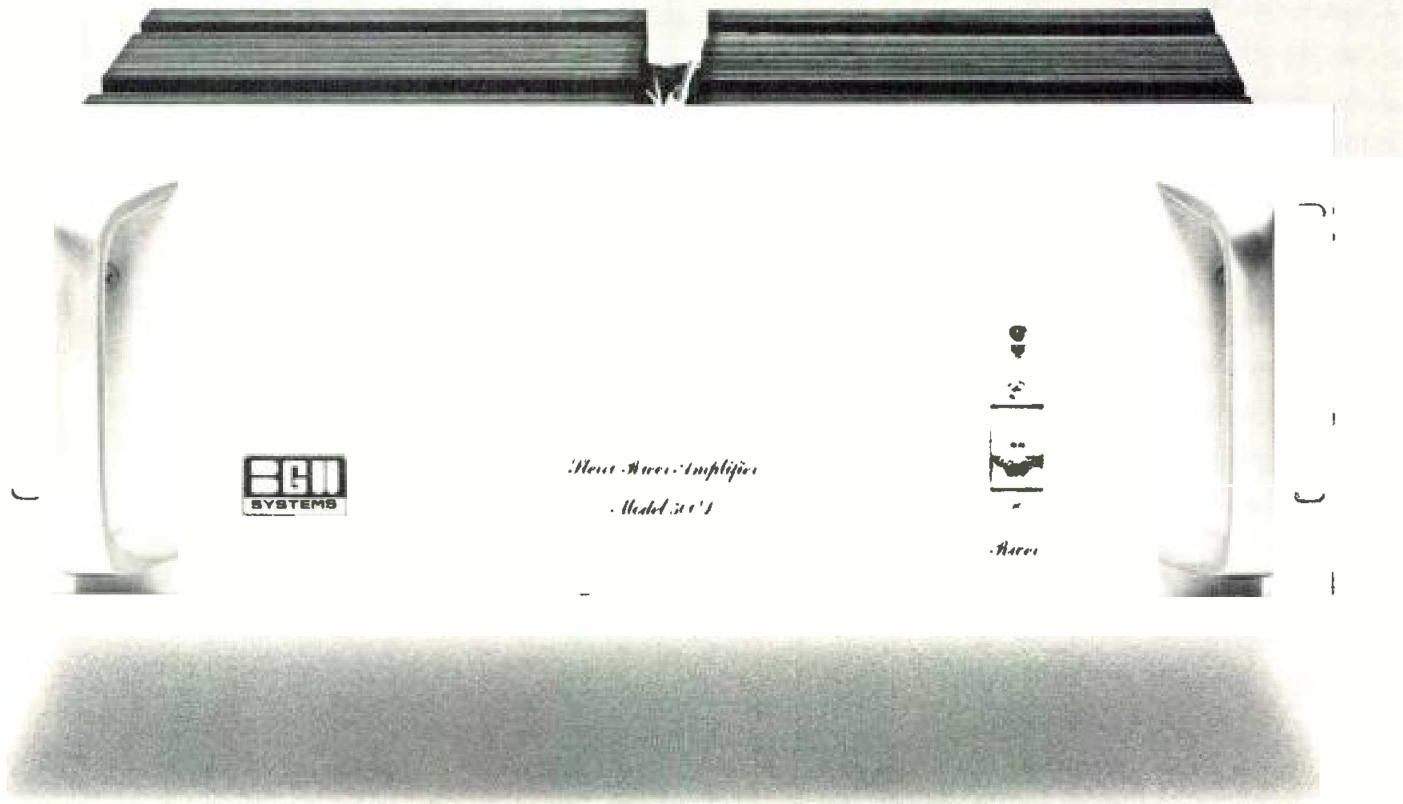

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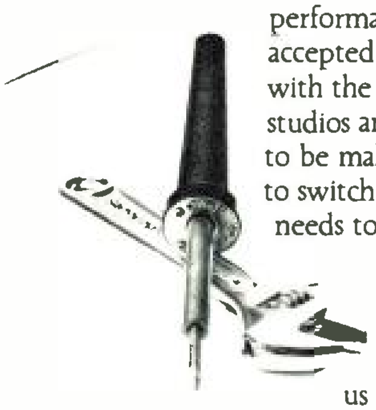

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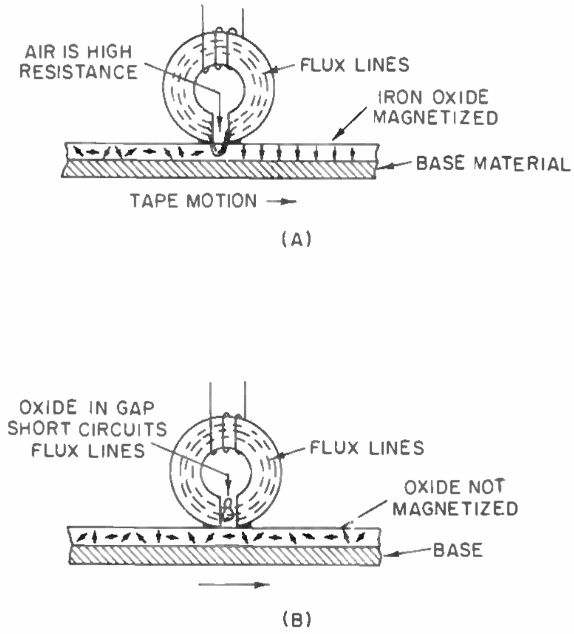


Figure 5. (A) The air gap has a high resistance to magnetic flux lines of the head and forces them through the easier path in the tape's iron oxide. (B) When oxide clogs the gap, the flux lines are short-circuited and few, if any, enter the tape.

Regular cleaning of all the machines will do much to maintain the technical quality of the end product.

TAPE

All audio tape is not alike even though it may visually appear the same. Various *types* of tape (not necessarily brands) have other differences besides thickness of the base material. The iron oxide coating differs. This is where the action is—at the iron oxide coating. The recorder should be optimized by adjusting the bias for the highest output and frequency response on a particular type of tape. If possible, the news room should standardize on one type. If there must be several types in use regularly, a compromise is necessary. Find a compromise setting of the bias and equalizers that will produce similar, but satisfactory results on all the types of tape in use. Remember that this is a compromise and therefore less than optimum, but it is better than having optimized results on one type and much less than optimum on all the other types. However, your best bet is to standardize on one type of tape.

BULK ERASING

Cartridges must be bulk erased, but it is often also advantageous to bulk erase small open reel tapes that will be taken out on portable recorders.

These machines are half-track and there may be occasions when an important story is brought back to the station with no time to dub it over on the master machine. The tape can be played directly on the control room machines (assuming it was run at the correct speed). Since most control room machines are full track, the half-track tape you are using must have the other track blank.

When bulk erasing tape, move the tape out of the eraser field before turning the eraser off. That strong collapsing field can severely magnetize the tape, creating a clicking or popping in the background of new recordings on that tape, something difficult to erase.

BATTERIES

Portable recorders will operate on batteries, usually the rechargeable type. It is important that the batteries be kept at full charge so the recorder is ready to use. If the batteries are down and the recorder taken out on assignment, an important interview may be lost because the batteries went dead. If they're not quite dead, just low enough so that the machine runs at a slow speed, when the tape is played back on a machine that is up to full speed, the results will be "Donald Duck" sounds. If the tape is open reel, it is sometimes possible to save the recording, provided that there is a machine which has a variable speed arrangement, such as a variable cue speed. Play the tape and adjust the speed control until the voice sounds natural. Once you can achieve this, then dub the results onto another tape, rather than rely on this arrangement. The best practice, of course, is a regular check on the batteries and the chargers.

TELCO RECORDING

Many interviews are done over the regular telephone, and the interview recorded. To get the best recording, connect directly to the telephone line for that phone number. But use 0.1 μFd capacitors in each side of the line for d.c. isolation and an audio transformer for a.c. isolation to prevent hum. Voice levels at the news room and the far end of the line will almost always have a wide difference. Use of a *gated* agc amplifier in the lineup will improve the recordings considerably to equalize the voice levels.

SUMMARY

The news room is a specialized, but important recording center of the station. Good technical quality requires attention to the impedances and signal levels in the mixing arrangement, and then a regular cleaning and maintenance program of the equipment. ■

db theory & practice

NORMAN H. CROWHURST

● The theory of a crossover is that at its design frequency both units will receive precisely 3 dB less signal than they receive in their pass range, so that each delivers exactly half the power at that frequency. As we have been saying, that theory is based on the assumption that the loudspeaker has an impedance that looks like a pure resistance, to receive that half of the power.

Whatever may be the situation with the unit that receives the high-pass ouput. the one that receives the low-pass output usually has an inductive impedance at this frequency. In the August issue, I showed that a low frequency (woofer) unit has an impedance that can be approximated as a combination of resistance and inductance. So you count the inductance as part of the crossover filter you design. That is the idea.

WOOFER INDUCTANCE

But to do that, you need to know the equivalent inductance of the woofer unit. How do you arrive at that? I have heard engineers argue that the value is the voice coil inductance of the unit. So they clamp the voice coil to stop it from moving and measure its inductance.

That might work out, if that was the way you operated your loudspeaker. But the speaker doesn't radiate very much sound with its voice coil clamped! You need to know the inductance component when the unit is working, in exactly the situation it will be when the whole system is operating.

As I said before, part of the equivalent inductance is due to the actual voice coil inductance and part of it is due to mass effects of the moving cone, and of air that the cone pushes around. Radiated energy appears principally as resistance, which adds to the voice coil resistance.

So how do you measure the effective inductive and resistive components, and at what frequency do you take them? The answer to the last part of that question should be obvious: at crossover. But a surprising number of people will use an a.c. bridge, which has its own built-in frequency, meaning that they measure it at that frequency. thus assuming that the values are constant. which they seldom are.

Anyway, you probably do not need a value as precise as one that a bridge can give you. So using a bridge to measure the unit's inductance will give you a value that is unnecessarily precise and, more importantly, wrong for the purpose you have in mind, unless the bridge measurement frequency just happens to be the same as the crossover you plan to use.

With a little care in calibration, those same ellipses I introduced to you in the last column can tell you what you want to know in order to design an appropriate crossover filter.

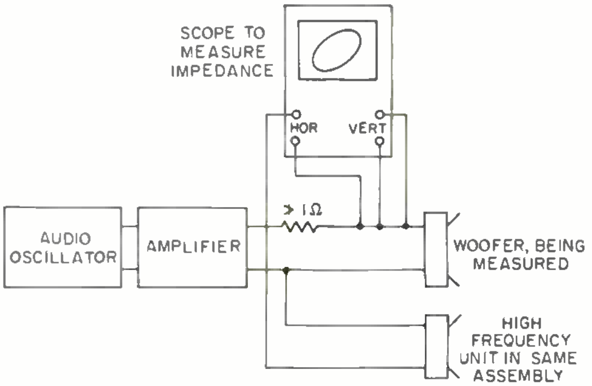
TESTING

First, you should make the test with the unit in position and, if possible, with the unit that takes high pass drive also connected. Be careful to measure only the impedance of the low-pass fed unit (FIGURE 1). This will produce a loading on the low-pass unit equivalent to what will exist when your design is complete. You may want to try reversing connections to the high-frequency unit to make sure the phasing is correct at crossover frequency.

To take your readings on the 'scope, use as small a resistor as possible in series, to measure current. that will give you an adequate horizontal deflection—not more than 1 ohm. You want the electrical damping of the unit to be as near operating value as possible.

You calibrate the vertical and horizontal scales in a conventional way so you know what they mean in voltage

Fig. 1. Schematic for oscilloscope set-up for measuring impedance of low-frequency unit of a crossover pair, with both units being driven in frequencies within range of crossover frequency to be chosen.



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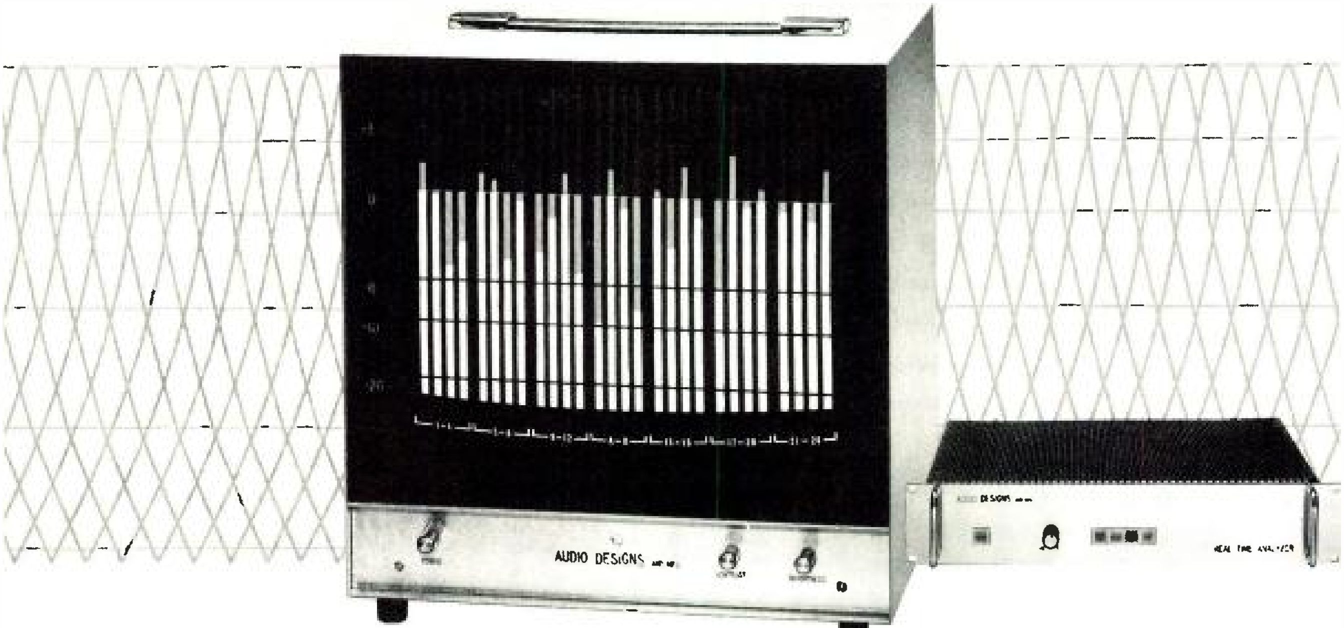

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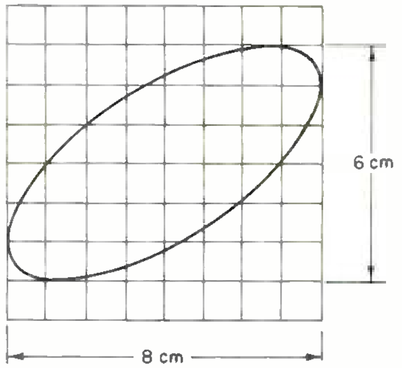


Fig. 2. Typical ellipse obtained to use as measure of magnitude of impedance, showing relevant measures for this purpose.

and current, respectively. Then the width and height of the ellipse will give you the impedance magnitude and the shape of the ellipse will tell you its phase.

This is how you do it. Suppose, with the gain of both defections calibrated, the width is 8 cm. and the height is 6 cm. The horizontal gain is set at 10 millivolts per centimeter, and your series resistance is 1 ohm.

8 cm. represents 80 millivolts across the 1 ohm, or 80 milliamps. If the vertical gain is set for 0.5 volt per centimeter, 6 cm. represents 3 volts. Both those values are peak-to-peak, by the way, so you do not need to convert them to rms because the ratios will be the same.

The impedance represented by this trace is one that will pass 80 milliamps with 3 volts across it, which figures to 3 ÷ 0.08 = 37.5 ohms. Now, to measure phase, you will find it easier to set the gain so height equals width. This will lose your calibration for impedance magnitude, but you already have that.

If the ellipse is skinny, the easiest way to figure phase is to take the intercepts on the axes (FIGURE 3). If you have set both height and width to 8 cm., and the average intercept on the axes is 1.5 cm. from the center of the grid, the sine of the phase angle is 1.5 ÷ 4 = 0.375. A table, or your pocket computer, will give the angle at 22 degrees.

Now, the resistance component is 37.5 ohms, times cos 22 degrees, which figures to 34.77 ohms. The reactance component is 37.5 ohms, times sine 22 degrees, which figures to about 5.27 ohms. Now, what you want is inductance, so you apply the formula $X_L = \omega L$, or $L = X_L \div \omega$, where ω is $2\pi f$.

With a frequency of 800 hertz, ω is 6.28 x 800 = 5024. And 5.27 ohms

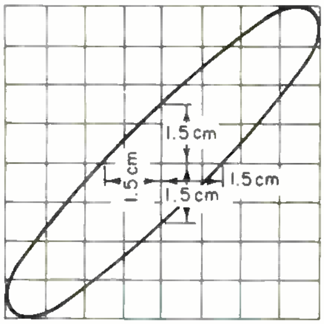


Fig. 3. Typical measurement of phase on a relatively narrow ellipse.

divided by 5.024 makes about 1.05 millihenries.

If the ellipse is broad, the easier way is to see how far from the axes the ellipse makes a tangent with the top, bottom and sides (FIGURE 4). This will give the cosine of the angle. If it is 1.5 cm. again, the cosine is 0.375, leading to the 68 degree angle, from which point the method will be the same, although the values will be different.

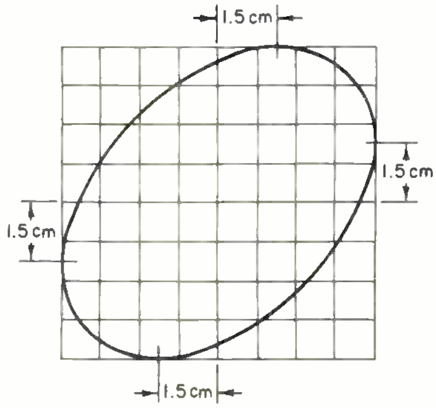
That method will enable you to compute resistance and inductance at just one frequency at a time. If you compute it at a number of frequencies, all near to crossover frequency since that is where you are concerned with the value, you may find the value varies a little. This is because the equivalent circuit is not quite as simple as just one resistance and inductance combined.

PLOTTING ON GRAPH PAPER

Perhaps an even easier way, and one that is accurate enough for most purposes, is to plot the impedance on logarithmic graph paper (FIGURE 5.) I have shown a probable impedance curve in solid line against which I have drawn a dashed line that could represent a simple resistance and inductance combination, showing the construction.

You will want to operate your low frequency unit with a cut off below

Fig. 4. Typical measurement of phase on a relatively broad ellipse.



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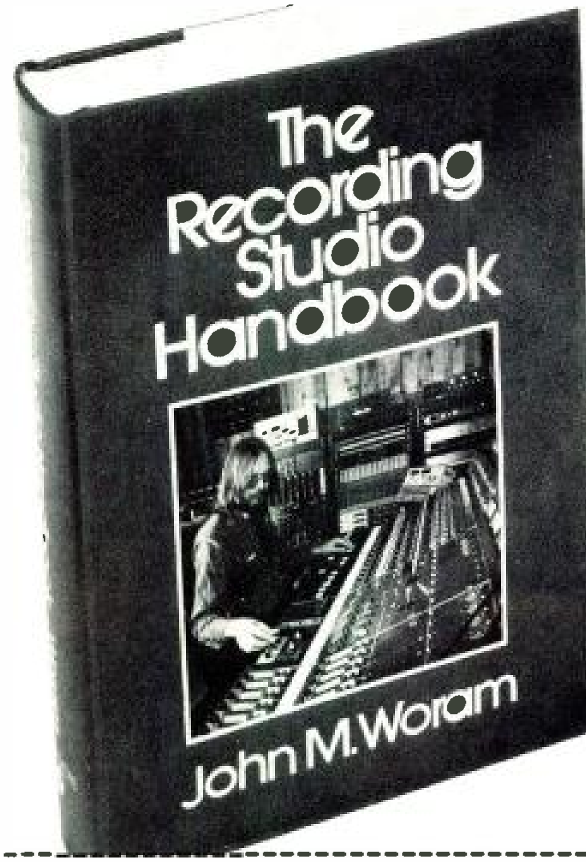

The technique of creative sound recording has never been more complex than it is today. The proliferation of new devices and techniques require the recording engineer to operate on a level of creativity somewhere between a technical superman and a virtuoso knob-twirler. This is a difficult and challenging road. But John Woram's new book will chart the way.

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John Woram is the former Eastern vice president of the Audio Engineering Society, and was a recording engineer at RCA and Chief Engineer at Vanguard Recording Society. He is now president of Woram Audio Associates.

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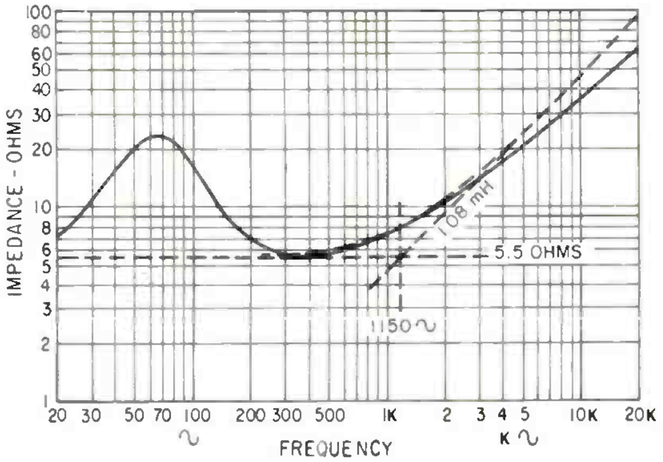

Fig. 5. An example of how to estimate resistance and inductance directly from a plot of impedance against frequency, using log-log paper. Note the idealization necessary to make the computation.

the point where its impedance, thus approximated by an ideal representation, is 1.414 times the resistance component. You can figure a workable value for the equivalent inductance from the frequency at which this 1.414 point occurs.

Suppose the best approximation of resistance is 5.5 ohms, and the frequency where the idealized impedance curve is 1.414 times 5.5, or 7.8 ohms (as nearly as you can read the graph). is 1,150∼. The value for ω is 6.28 times this, or 7,222. Dividing 7.8 ohms by 7,222 gives 1.08 millihenries as the equivalent inductance.

A suitable crossover frequency would be anything below 1,000 hertz, although not too far below.

Now, to complete a crossover design. you need to pick a configuration that has an inductance in the output lead of the low-pass section so that tht loudspeaker's equivalent inductance can be part of it. Suppose the crossover design, for the impedance and frequency chosen, calls for 1.6 millihenries. Then you will use an actual value that makes it up, with whatever you figure the loudspeaker's inductance to be. If it was 1.08 millihenries, your inductor in the network needs to be 1.6 − 1.08 = 0.52 millihenries, or 520 microhenries.

You should design your network for the resistive impedance of the loudspeaker. If this is 5.5 ohms, then design your network for that value, for best results. It may be nominally an 8-ohm loudspeaker, and perhaps has values that straddle that nominal value throughout its pass range. The amplifier should use an 8-ohm output, that is, one designed to feed an 8-ohm speaker. Except near crossover, the network will pass the actual loudspeaker's impedance back to the amplifier, unchanged in value.

The amplifier may be a little mismatched at crossover frequency, but the filter will assure the correct shape of roll-off for the design chosen, that is, how many dB per octave you choose to use. ■

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MARTIN DICKSTEIN

• In the past three columns, this department has discussed the different phases of putting on a presentation, from the technician's side. After the first column was written, this topic was discussed with an executive of a large company which made presentations regularly to employees, other executives of the company from overseas, and clients. Our friend suggested that it might be a good idea to discuss briefly in more detail a few of the more critical items that would help make a better setup, a more expedient operation, and a faster tear-down. Here are a few which were *not* mentioned in the previous columns, and one or two I did.

FILM PROJECTOR

First, a quick look at the film projector. The choice, in general, is between a manual load and an autoload model. The decision depends to a great extent on the film to be shown, and how adept the projectionist is in handling an emergency. If the films are all on one reel, and the film can be set up before the meeeting gets under way, either projector can be used. With the manual type, the film can be threaded, focused, the sound level set, and run back to the beginning. With an autoload, the front, or head end, must be clipped straight across to allow it to thread smoothly. Most projectors come with a cutter to take care of this. Once the film has begun to run, is focused, and the sound set, it, too, can be reset to the start. (With a manual projector, of course, the switch must be put back in the forward position.)

The problem of choice takes on a different appearance when one looks at the mechanics of the projectors. If the person running the films knows how to thread a manual projector easily and properly, there is no problem. If not, the film can come loose, lose a loop and flutter on the screen, or cause warbling sound. A good technician knows how to handle this problem. Normally, this won't happen if care is taken and the film tried before the show starts, but if there is a bad sprocket hole which doesn't show up until the first play begins, it might lose the loop. When this occurs, it is safer to stop the projector momentarily, fix the loop quickly, and resume. Some projectionists like to adjust the film while it is running, with a quick finger touch. This method is quicker, certainly, and looks better, but don't try it yourself unless you know what you're doing. Otherwise . . .

A problem could also arise if only half the film is played or if the film breaks in the middle. With a manual projector, the technician removes the film (provided he or she knows how) quickly, runs it past the break, adds a piece of tape to hook one end to the other after the sprockets, and continues the film. On an autoload, this quick patchwork takes a bit more work. Since the machine is made for loading film automatically, the path is built to prevent the film from slipping out accidentally. It is this same safeguard that makes it more difficult to remove the film before the reel is finished.

A good technician might know to remove the cover over the exciter lamp, to open the sprocket covers, etc.—it just takes more time. When the film is spliced together, there are other complications. The autoload machine is made to load from the front, and from the head end of the film. Quite a bit of know-how is necessary to rethread the machine manually. It might be easier to feed in the supply end and let the reel run through before the splice is made. However, cutting a straight end could lose a frame or more of the film picture and sound. Tricky, eh?

One more problem to think about, as far as choice is concerned, comes with knowing there will be more than one film. The first can be put on at the outset. How does one handle the others? With a manual projector, there is no problem. The first film is removed, another take-up reel put on, the second film threaded and cued up (quietly) and the show goes on.

When there is self-threader, however, the film's leader end must go through before the first frame is in position. If the machine is in the same room as the meeting, the manual unit gets the definite nod. It's easier to handle for one who is familiar with it. (Incidentally, it also helps to know how long the films are and how many so the proper takeup reels can be provided, both for size and quantity.)

SLIDE PROJECTOR

When it comes to the slide projector, there are also some things to think about. If the slides will fit into one standard universal drum (assuming the projector to be the carousel type), the greatest safety is achieved with the grey drum. It will permit use of metal and plastic mounts as well as cardboard. The black carousel and the 140's are intended for thin plastic or cardboard respectively and will not handle the thicker mounts. The standard types of projectors will also be less of a problem, usually, even with cardboard, in the event one corner of the mount lifts off or curls slightly. The wider spacing permits a bit of leeway, The thinner slits do not, and a poorly mounted or bent cardboard will jam.

It is also better to use the autofocus models so that slides will focus automatically. When cardboard and glass mounts are mixed, there is sometimes (usually) a variance in the focus. If the presenter does not mind adjusting the focus manually from his remote control, fine; then it makes no difference in the choice of mount, or machine. It is much smoother, most times, however, for the machine to focus automatically. There is also a model and a control which gives the presenter the option of manual override even with an autofocus. This could be the best, yet. (Incidentally, glass mounts, either in metal or plastic, keep the slides from curling as they are sometimes wont to do in cardboard, and will protect the film from finger handling.)

SCREENS

The location of the screen and the shape of the room make a big difference. If the room is narrow and long, a beaded screen is probably good. This type gives a very bright reflection, but in a narrow angle. The audience should normally sit no farther than 25 degrees to either side of the center line. Beyond this, the intensity falls off. (You will notice a similar effect, although at a different angle, with a rear screen, where the slide seems uniformly bright as you sit in the center, but falls off on the side opposite as you're moving either to left or right of center.)

With a lenticular or Ektalite screen, the angle can be spread to about 30 degrees at either side of center. Using matté material, however, allows for about 45 degrees on both sides of center, and thus offers the widest seating spread of all. This means that (with the front row about twice the width of the image away from the screen, and the last row about six times away) the front row should only be as wide as about two image widths for a beaded screen and the back row about six times as wide. For a matté screen, the front row can be as wide as about three times the width of the image, and the last row as wide as twelve images.

LENSES

The simplest though not very practical, way to get the right lens for a projector is to take a series of them with you when you set up. For the slide projector, for example, you could take a 3 in., a 5 in., a zoom to cover 4 in. to 6 in., a 7 in., and a 9 in. size. There are also 8 in., 8½ in., and other lenses available. To be safe, you might get all of them just to have "in case." If you do the same with your film projector, your supply of lenses will take up more room than your equipment!

In most instances, the zoom and the 7 in. for the slide unit as well as the zoom and 2 in. for the film projector will suffice. If you will look at available charts you will find that for throw distances of 13 ft. to 94 ft., the 7 in. slide lens matches the 2 in. film lens, considering the requirements for screen width. Thus, if you project 2 x 2 in. slides from a distance that will fill a screen width, using a 7 in. lens, the film will provide a similar size image with a 2 in. lens.

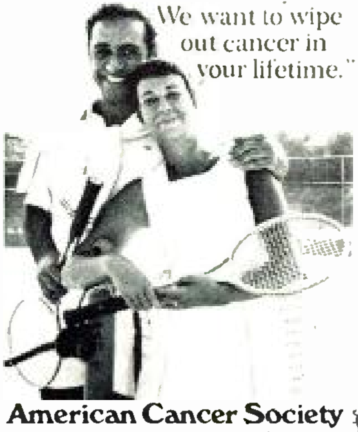
Where the size of the screen or the projection distance are such that a different size lens is required, trial and error on the site might be necessary until the projectors are positioned properly. However, it might help to determine beforehand what range of lenses, at least, you might need.

The formula says to multiply the projector aperture width in inches by the projection distance in feet, and then divide by the lens focal length in inches to arrive at the screen width in feet. To put this a different way, the lens in inches can be found by multiplying the aperture width (inches) by the throw distance (feet) and dividing by the screen width (feet). For the 16mm film projector, the aperture width is 0.38 in., and it's 1.3 in. for the slide projector. Using this calculation, you will be very close to the figure you want to determine the screen width you should have, the lens you'll need, or the throw distance you require for the equipment you will use. Remember that the longer the lens, the smaller the image will be from the same location, or the farther back you can go for the same size image. This reminds me of a poem I found a long time ago but can't remember where (apologies to the publication) written by Donald Canty:

When you want to say a lot,
Put a short lens on the shot.
If you want to frame it tight,
Use the longest lens in sight.
If your actors move about,
Now in focus, now they're out.
To a shorter lens you yield
To increase your depth of field.
If the object seems too small,
Or you see far too much wall,
Then this lesson you must learn,
To a longer lens you turn.

The poem is actually aimed at photographers, but parts hold true as well for projection.

Using these hints as guidelines, it might be a little easier to figure out what equipment to purchase, rent, or take out of stock for a presentation, what screen to choose for material and size, what lenses to take, and the extras you can find helpful to set up, run, and take down a show as if you really know what you're doing . . . professionally. ■



db the sync track

JOHN M. WORAM

• *An automated editorial or, "We'll fix it in the computerized mix."*

Those who proclaim that technology is slowly ruining the art of recording should have a field day with this issue of **db**. Each of the three automation systems described within these pages brings computer technology into the studio control room, where it may be used, or abused, depending on who's in charge.

Each system has its distinctive operational features. Necam gives us servo-driven faders, for instinctive update. Allison Research suggests a master control concept that eliminates the usual redundancy of console controls. Automated Processes uses its own time code, and stores instructions on a data cartridge.

Each system is totally incompatible with the others, although this may not be quite the disaster that it seems at first. So far, automation remains pretty much a mixdown tool, and usually mixdown work is confined to one *venue* anyway. An occasional feature might be useful during recording, such as instant reset of all equalizers, but this is not the sort of thing that needs to be carried from one studio to another.

VERSATILITY

Automation gives the engineer an incredibly versatile production tool. After a few mixes have been stored in memory they can be electronically edited into any conceivable sequence without lifting a razor blade. This feature can be used to advantage in classical music production work as well as in the more obvious multi-track applications.

In the former, each take may be assigned to memory, and when the artist or conductor decides he'd like to hear several different edited versions of the performance, he may be obliged instantly. If the system is available during recording, this sort of instant editing may even be done during playbacks, thus saving a lot of educated guesswork about how it will all sound later on.

During any mixdown session, there is an obvious saving in two-track tape, since there is really no point in running the two-track recorder until the ideal mix has been worked out. And, if you are concerned about wearing out the multi-track master tape with repeated run-throughs, just make a work copy and use that until you've got your mix worked out. Then return to the master, press the button and go. If the two-track master sounds a bit bright, it could be that all those rough mixes gradually wore down the high end of the work copy, while the still-fresh master retains the original response. At least one manufacturer is studying this possibility, with an eye towards some sort of built-in automated correction factor.

Do these and all the other automated advantages mean that recordings will get better and better? The answer to that one is a definite yes and no. (And remember you read it here first!)

THE HUMAN INGREDIENT

Automation is just one more tool to be placed in the hands of the engineer or producer. But—science fiction notwithstanding—it doesn't have much brain power. That's one optional accessory that you have to supply yourself. If you don't understand the technology you're using, your production will surely suffer. But of course, that's as true of the simplest microphone as of the most sophisticated automation package.

By now, even pre-automation technology has progressed to the point where many people are getting left behind. An interview in *Billboard Magazine* some time ago is a good—but depressing—example. An unidentified "industry spokesman" commented on recording equipment. "The real problem," he says, "is that this equipment is made by people who can't evaluate it from a musical viewpoint. And this is not only the people making consoles. This goes for microphone manufacturers and others you may not think of." Of recordings, he observes. "Noise-wise, the current ones are better. but from a musical reproduction, the older ones (20 years ago) win out."

Unfortunately, this sort of reverse logic is very prevalent in the industry. It goes like this:

Premise 1: Recordings of twenty years ago sound better than many of today's releases.

Premise 2: Technology has proliferated over the last twenty years.

Conclusion: Technology is ruining recording.

The more imaginative engineer will object to this sort of reasoning. He may point out that with the technology available today, it is possible to make recordings that are far better than anything produced even ten years ago. But he also knows that it is possible to demolish almost any performance by treating it as a sort of musical Heathkit, to be assembled after hours by a knob-crazed producer (or engineer). So, let's re-write that proposition:

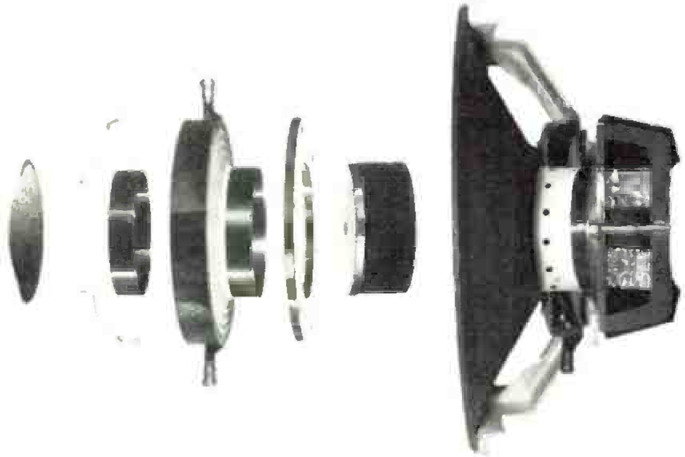
Premise 1: Recordings of twenty years ago often sound superb when heard today.

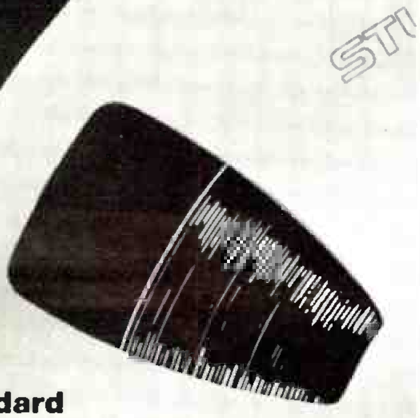
Premise 2: Technology has proliferated over the last twenty years.

Conclusion: Today's recordings *could* sound even better than the older ones.

The key word is "could." The industry spokesman quoted above is distressed by what he hears, and blames technology. He forgets that *he* is still in charge though, and if he doesn't like what he hears, it's probably because he is mis-using the tools that are available to him. Give him a little automation, and he's really going to get upset!

Automation is not unlike atomic power; it can be used creatively or destructively, though perhaps the results are not quite so dramatic. Used creatively, by knowledgeable engineers and producers, there's no reason why it cannot become one of the most powerful tools in the advancement of the science—AND art—of recording. ■





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Mfr: Crown International
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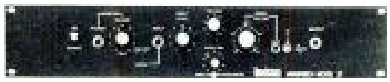

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Mfr: Lexicon, Inc.
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Mfr: Russound
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Mfr: ESS, Inc.
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● Stereo four-band parametric equalizer SC-66 may be set sharply enough to equalize an individual musical note. Claimed distortion is less than 0.05 per cent and noise −87 dBV. Four overlapping bands cover the entire audible range (16 Hz to 23 kHz). The power supply is self-contained. Applications include feedback control, acoustical correction, improvement of mic and speaker response, and generation of special effects.
Mfr: Ashly Audio, Inc.
Price: $599.00.
Circle 55 on Reader Service Card

PORTABLE MIXING SYSTEMS

● Four new mixing systems have recently been introduced by this manufacturer. The M 82 line includes M 82-1 Musicians' Mixer system, M 82-II VCA Stereo Mixing system, B 82-B Broadcast system, and M 82-C Consultants' system. Each system has eight master input channels with optional 4-input sub master additions, providing a maximum channel complement of 24. The units feature a foldback system and separate monitors. The lightweight mixers, constructed of 14 gauge steel, weigh less than 26 pounds.

Mfr: EDCOR

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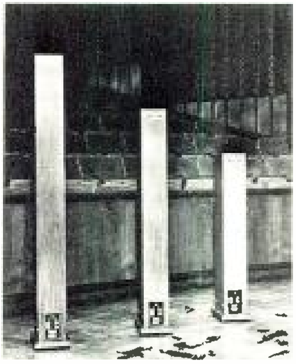
REVERBERATION CHAMBERS



● Extended bass response is notable in the MasterRoom reverberation chambers. Normally rolled off at 6 dB per octave below 200 Hz, extended bass response units carry the flat response to 50 Hz and are less than 8 dB down at 20 Hz. Units with this characteristic are the same price as the rest of the manufacturer's line. Kits may be purchased to convert existing units to the extended bass capacity.

Mfr: MICMIX Audio Products, Inc.

Circle 52 on Reader Service Card

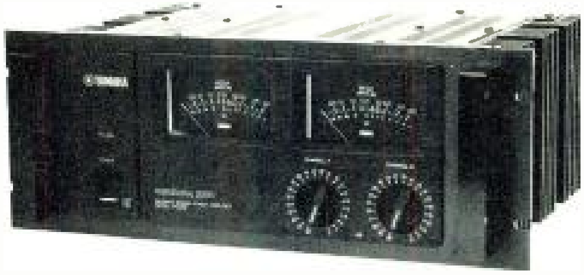
MULTIBAND AUDIO PROCESSOR

● Eight independent bands of compression to increase carrier modulation to a figure approaching theoretical maximum are present in Model 230 multiband audio processor, designed for a.m. and f.m. radio broadcast use. Each of the eight bands has individual threshold and compression adjustments, permitting response shaping to complement the programming format. A final peak limiter contains program peaks within absolute prescribed limits. Other features include gated expansion, program-controlled phase inversion, and separate frequency-selective limiter for f.m., with both 25- and 75-microsecond characteristics.

Mfr: Inovonics, Inc.

Circle 53 on Reader Service Card

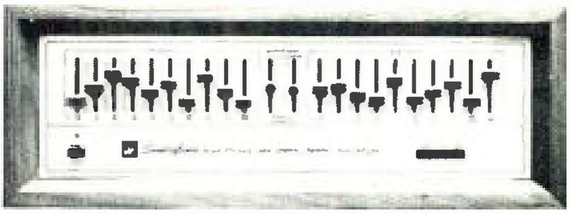
HIGH POWER AMPLIFIER



• High power handling recommends P-2200 amplifier for disco use, while the precision of the peak-reading meters, which display five decades (50 dB) of output level, suits the unit for studio monitoring. Less than 0.5 per cent thd is claimed at full rated power. The peak meters have illuminated faces marked with dB and watts-into-8-ohms scales (the unit packs 4-ohms sustained output at 350 watts). Input attentuators are marked in 22 calibrated dB steps, detented for extra accuracy. Other features include input connectors for both channels, including one male and one female XLR connector, a parallel phone jack, a polarity switch allowing either pin 2 or pin 3 to be selected. Outputs use 5-way binding posts which give the choice of direct-wired connections or high-current banana plugs.

Mfr: Yamaha International Corp.
Circle 56 on Reader Service Card

TAPE-PLAYBACK EQUALIZER



• Featuring octave-by-octave graphic equalization, RP2204 equalizer is designed for connection to the tape monitor circuit of any stereo receiver or preamplifier. It includes its own tape monitor inputs and outputs with front panel pushbutton selection, which also provides tape record equalization switching and automatic "equalizer-on" for environmental equalization (an environmental test record is included). The unit has two completely separate ten-octave equalization panels, with plus or minus 12 dB boost and cut provided for each channel, enabling exact balancing of input to output levels within an 18 dB range.

Mfr: Soundcraftsmen
Price: $329.50.
Circle 57 on Reader Service Card

A MASTER OF THE ART TALKS ABOUT MASTER-ROOM™

"We have used Master-Room reverberation with Stevie Wonder, Neil Diamond, Joe Walsh, Petula Clark, Rufus, Fleetwood Mac, Tower of Power, Billy Preston, Sergio Mendez, Isley Brothers, Ramsey Lewis and a number of others. In fact we have used our Master-Room chambers on virtually every record we've done here.

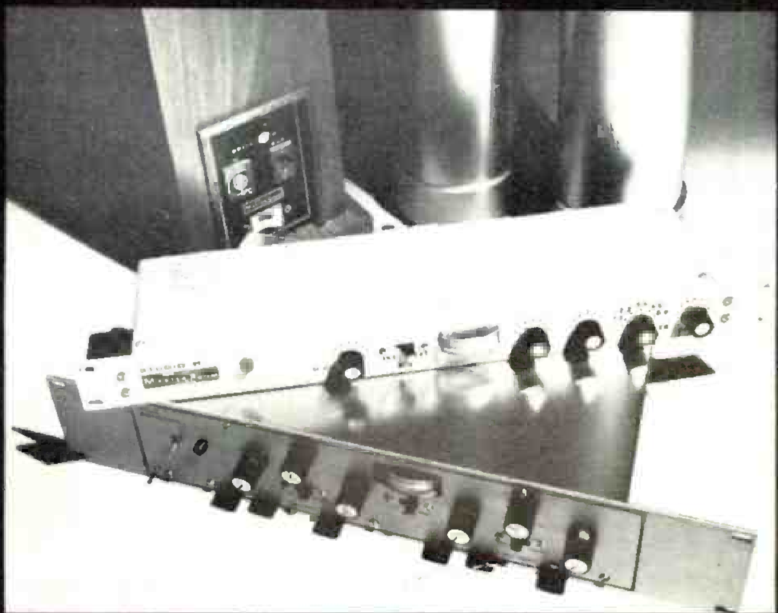
"What I like about Master-Room sound primarily is the relatively clean first order echo. It's a real smooth system.

"We like it especially on electronic instruments, and the Master-Room can even take drums and sound good. It's much better than any other comparable type reverb and costs much less.

"Here at Kendun Recorders we have unlimited funds for equipment and we can afford the best. So, along with our live chambers, there are two or three Master-Room units in each of my rooms.

"I would feel incomplete without my Master-Room chambers."

Kent Duncan
President
Kendun Recorders
Burbank, California



Master-Room reverberation chambers alone won't make you a Master of the Art, but they will provide you with the very finest in Natural Sound Ambience.

Models available to suit every need. See your dealer or write direct.

Circle 42 on Reader Service Card

MICMIX AUDIO PRODUCTS, INC.
9990 MONROE DRIVE, SUITE 222 • DALLAS, TEXAS 75220
(214) 352-3811

A sine wave time code minimizes crosstalk and enables locational information to be read at high speed, even with the tape lifted off the head.

JOHN WORAM

A Look at Automix

As many readers may know by now, when a 16-track studio becomes automated, it simultaneously becomes either a 14- or a 15-track studio, depending on the automation system used. Some require two tracks for the automation instructions, while others put a SMPTE time code on one track, with a synchronized floppy disc used for data storage.

In developing its current "Automix" system, Automated Processes has chosen a variation on the latter technique, using the company's own time code instead of SMPTE's. Automated's Carl DeWilde explains that the SMPTE code—essentially a square wave—has a tendency towards cross-talk, and may often be readily heard on adjacent audio tracks. With the tape recorder in fast forward or rewind, the code's playback frequency is so high that a special wide band amplifier is required, and the tape lift mechanism must be defeated in order to read it, with this in turn causing additional head wear.

John Woram, in addition to being a regular **db** *columnist, is an audio consultant, an instructor in the New York Institute of Technology's graduate program, and, most prestigiously, author of the newly published RECORDING STUDIO HANDBOOK.*

The Automated Processes time code is modulated on a 3 kHz carrier, and looks like a sine wave, with no more cross-talk than any audio signal. In addition, there is a 30 Hz carrier which enables location information to be read at high speed, even with the tape lifted off the head.

FIRST MIXDOWN

Automated's data storage medium may be easier to use than to explain, but here goes anyway. During a first mixdown, the console is scanned 37.5 times per second, and the analog VCA levels are converted to digital information and stored in one of three "cache memories" within the programmer. This stored information—in increments corresponding to about 1.7 seconds of real time—is then transferred to the data storage medium (more on this in a minute) in a fraction of a second, while the programmer is storing the next increment of information, which will then be transferred in the same manner.

THE 3M DC 300A DATA CARTRIDGE

Automated Processes' data storage medium is the 3M DC 300A Data Cartridge, chosen because of its physical durability and a storage capacity that is about ten times greater than the floppy disc. DeWilde points out that the cartridge does not run continuously but rather in short spurts, during which the data is either written (recorded) on, or read (played back) from, the cartridge. Since these brief spurts take place in a fraction of a second, and transfer almost two seconds' worth of instructions, there is plenty of time for the data tape to shuttle ahead or backwards, while the programmer is using up the previously stored 1.7 seconds' worth of information.

On the odd chance that this explanation is not crystal clear, consider the following typical situation. At t=0, a mixdown is begun. For the first 1.7 seconds, the data cartridge does not move; however the programmer is storing instructions in one of its three cache memories. At t=1.7 seconds this information is entered on the data tape. The transfer takes a fraction of a second. Meanwhile, the next 1.7 seconds' worth of instructions (t=1.7—3.4 seconds) is being stored in another memory. At t=3.4 seconds, this information will be transferred to the data tape, and so on.

During playback, the data cartridge transfers the first 1.7 seconds of stored instructions back into the programmer, and from there it is converted back to analog and routed to the VCAs in the console, while the time code—which has been stored and retrieved along with the mixing instructions—keeps everything in sync.

To "update" the mix, new instructions are entered on the data cartridge. Within each "data block" of 1.7 seconds, there is room for ten different sets of mixing instructions, although during routine operation only two sets will be stored. These are known as "current" and "previous."

To get through this part of the system, let's look at another typical situation. The session begins with a series of rough mixes, which we'll call A, B, C and so forth. The first few mixes aren't worth hearing, and so we continue mixing without wasting time listening to playbacks. However, at the completion of mix F, we decide to have a listen by simply playing back the multi-track tape, the "current" mix (F) will be heard. We may also hear the previous mix (E) by depressing the "previous" button. However, all earlier mixes have been lost. (Remember that up to now there has been no reason to do any actual recording onto a two-track tape)

If we hear nothing worth saving, and go back to mixing, then as mix G is being done, F becomes "previous" and E is lost.

Sooner or later, there will be a mix worth saving. Let's say that R shows promise, but that S is a disaster. Before going on to T, we depress the following sequence of buttons: "previous," "store," "1." Mix R is now stored in Memory 1. Now, as mix T begins, R is replaced by S in the "previous" slot, but is retained in Memory 1 for future recall. Seven other mixes may be stored in Memories 2 through 8.

As the evening wears on, and the memories fill up, we may discover that the ideal mix seems to consist of the beginning of Memory 5, the middle of 3, and the end of 8. No problem. We simply play back the tape, depressing Memory buttons 5, 3 and 8 at the appropriate moments. If the changes are particularly tight, the tape may be stopped and rocked into position—as in editing—but in most cases this type of compositing may be done while the tape is playing.

This composite mix is now the current one, and as we listen we discover we can use a bit more drum sound here and there. Again, no problem. We can update the drums as we go along. The composite 5, 3, 8 becomes "previous" and the corrected composite becomes "current." As you may suspect by now, we can store the composite 5, 3, 8 and/or the drum-corrected version in one or two unused memories and go on mixing and re-mixing, long into the night. Or we can unload the cartridge and come back again tomorrow to pick up the pieces. We don't even have to remember the equalizer settings, since the data cartridge attends to that for us.

THE AUTOMATED EQUALIZER

Unlike the Allison master equalizer concept, the Automated Processes board has a standard-looking three-knob/three-switch equalizer in each module. During mixdown, the settings may be adjusted in the usual manner. However, if changes are made, then during playback there will be no similarity between the physical settings of the knobs and the actual equalization being heard. The knobs remain just as we left them, while the actual equalization settings correspond to the changes we made during the mix.

It is conceivable that we might be listening to a segment where, we recall, the equalization was say, −6 dB @ 7.5 kHz, although the equalizer knob/switch combination now shows perhaps +3 dB @ 10 kHz, since that's where it wound up by the end of the mixdown. If we wish to update the equalization, the knobs must be returned to the early setting first, or there will be a very sudden change as the equalizer leaps toward +3 dB @ 10 kHz. On the other hand, if we don't happen to recall the former setting, the equalizer may be "nulled" by depressing a pushbutton on the appropriate input module. This addresses a 6-led null indicator to the equalizer, and the leds indicate when each of the six equalizer controls has been matched in position to the actual equalization present at the moment. Once this has been done, a smooth update transition may be made.

The procedure is probably more of a chore in theory than in practice, since radical changes of equalization are usually not required within most routine mixes. If there is to be a significant change of settings, it probably happens during a pause or break of some sort. Therefore, updates probably do not require the nulling procedure anyway.

WHAT ABOUT RECORDING?

In most cases, the Automix console is not intended for use on involved multi-track recording sessions. Typically, the console will be equipped for overdubbing purposes, but will not have a microphone level input on every channel. In its standard configuration, the Automix board may have up to 32 inputs and 4 outputs. ■

Allison Research computerizes equalizers and faders, keeping the fancy hardware to a minimum.

JOHN WORAM

Paul Buff and the "Memory Plus"

AUTOMATION? The word usually conjures up visions of super consoles with faders sliding up and down, guided by unseen hands (see John Borwick's story in this issue). Above the console, a CRT terminal keeps the engineer in touch with what's happening, and nearby there is a keyboard for giving instructions to the computer, which may be down the hall somewhere. (Also down the hall somewhere, the maintenance man has locked himself in the bathroom, cowering in terror at the thought of an equipment breakdown.)

Allison Research was one of the early pioneers in the automation game, and their second generation hardware offers an interesting approach to the automated console. In explaining the system, Allison's Paul Buff points out that during even the most complicated non-automated sessions, relatively few functions are being changed at any given instant—a point that is not being overlooked by some of the competition's second generations either. The truth is that the overwhelming majority of the console's hundreds of buttons, knobs, switches and faders are—most of the time—lying untouched.

But with all of these controls available, it takes time for a computer system to scan a complex multi-function console, searching for chores to do, like gain riding, equalization changing, panning, echo-sending, and what-have-you. If the engineer is making just one or two changes out of a possible thousand or more, the computer is obviously wasting a lot of time, scanning all those static functions.

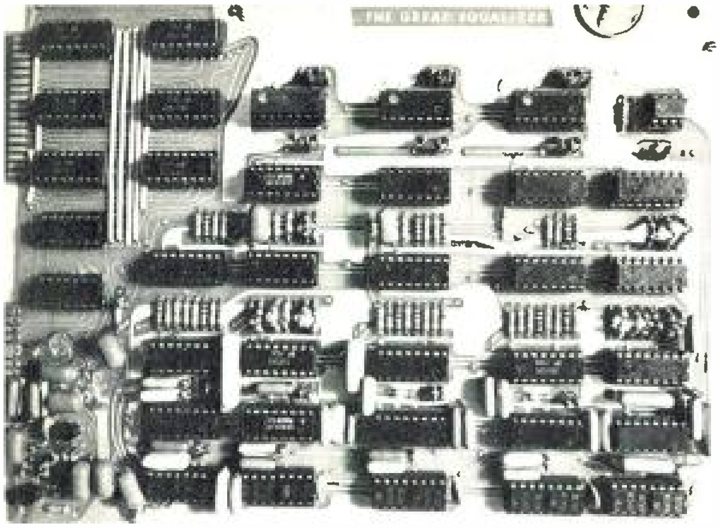
So, Allison Research's new Memory Plus does not scan the entire system, but rather responds only to function changes. Therefore, if the five drum tracks are just fine, the computer ignores them and concentrates where assistance is needed.

Unlike the earlier Memory's Little Helper system, which used a quinary (i.e. five possible states) system, Memory Plus is binary. Therefore, all information is represented by those familiar 1's and 0's, eliminating much of the ambiguity that can creep into more complex encoding systems. Now, the data takes the form of digital "words"—a series of rounded-off square waves, with a one represented by a 100 microsecond duration pulse, and a zero by 50 microseconds.

The first twelve pulses, or *bits* gives the "address" of some console function. Since each of the twelve bits can be either a 1 or a 0, we have the possibility of 2^{12} different addresses, or 4,096 separate functions. For example, the address of fader 17 could be 000000011011, leaving 4,095 addresses available for everything else.

We wondered if that wasn't a bit extravagant, since even an unlikely 48 in/48 out console with twenty separate controls in each of its input and output channels would require a mere 20 (48 + 48) = 1,920 addresses. However, the twelve-bit capability means that the console and system can easily accommodate a variety of different types of signal processing devices with a minimum of confusion. For example, a Brand X equalizer in channel 7 could have a certain address of its own. If a Brand Y equalizer was inserted in channel 7, it would have a different address. Therefore, instructions intended for one device would not wind up confusing another one.

The twelve bit address is followed by sixteen bits of



Allison's "The Great Equalizer."

Allison's "The Fabulous Fader."

data. For a fader address, the data consists of:

- 8 bits—level
- 1 bit —muting
- 1 bit —solo
- 2 bits—input assign (will assign 1 of 4 possible inputs)
- 4 bits—group assign (will assign to 1 of 16 group masters

For level control, eight bits = $2^8 = 256$. Therefore, the fader can be placed at any of 256 different levels. The Allison system covers the first 48 dB in 0.25 dB increments, and then goes to 1 dB increments, for a total dynamic range of 112 dB.

THE ALLISON EQUIPPED-CONSOLE

For the moment, Allison Research is building equalizers and faders only, with the total system to be put together by others. This should suggest all sorts of intriguing possibilities for the confirmed do-it-yourselfer.

For those who equate versatility with size, an Allison-equipped console may appear deceptively simple. To understand why, Mr. Buff has done away with most of the redundancy of knobs and things that clutter up the usual super-board. For example, consider an equalizer with the following functions:

High frequency cut-off—8 selectable frequencies, 18 dB/octave;
Three bands of equalization—8 selectable frequencies in each band ±15 dB peaking *or* shelving;
Low frequency cut-off—8 selectable frequencies, 18 dB/octave;
Equalization in/out switch;
Phase reversal switch.

In a conventional format, such an equalizer would certainly contribute its share of confusion to the general clutter. But Buff feels that the engineer spends comparatively little time looking at the equalizer, and when he does, the packing density makes it hard to "read out" anyway.

MASTER PANEL

As you may suspect by now, the Allison equalizer doesn't have a knob anywhere, as you can see in the picture. Instead, there will be a conveniently located master panel containing a series of momentary contact pushbuttons and a visual display of one channel's-worth of equalization functions. To set the equalizer in, say, input 3, just press a single pushbutton above fader 3. The master panel shows the equalization now in the channel 3 equalizer. Punch in some new e.q. settings on the panel and these instructions are instantly entered in the equalizer. To change the e.q. in another channel, just depress the pushbutton above the appropriate fader and repeat the process.

As an added bonus, the e.q. setting may be displayed on a cathode ray oscilloscope as a frequency response curve which changes as you vary the equalization setting. And when the session is over, a master clear button will remove all e.q. settings, across the board. Or if you like, you can retain the e.q. settings in memory and recall them later. More on this after we discuss the faders.

FADERS

The faders are different too. Perhaps our picture doesn't help much, but the fader is an optical counting device, using an endless belt. To operate, just slide your finger up and down the belt. The belt moves, and its opaque bars interrupt an optical path between an infra-red emitter and a pair of photo-detectors, and the audio level changes in 0.25 dB increments.

To let you know where you're at, a series of 32 leds shines through the belt, indicating the "position" of the fader, just as the old fader knob used to do. But unlike the knob, the belt can be touched—and therefore moved—from any point along its exposed path.

Now then, once you've established a basic starting point for each input level and equalization setting, this collective data may be stored on a conventional cassette tape by depressing the *load preset* button on the master panel. A series of presets can be stored, and then returned sequentially by depressing the *activate preset* button. This could make life on the road a lot easier. Each tune could have its own basic preset of balance and equalization stored, and the board could be set and re-set instantly with a minimum of hassle.

For the studio planning a gradual shift towards automation, Memory Plus may begin as a direct replacement of the existing faders, plus an external central controller. This will give the user automated level, mute, solo, group and input assignment capability. As with the equalization master panel—which can be added later on—there is a two way link between the faders and the central controller. For example, to "read" the faders that are assigned to group 15, just depress the group master 15 button at central control. An led at the top of each fader assigned to group 15 will light up. To assign a fader to any group, just depress the push-button above it, while holding down the appropriate group master button. Other buttons allow clearing individual functions or the entire console.

Memory Plus is not married to any particular data storage medium. So, depending on your needs, those ones and zeros can be stored on a floppy disc, an auxiliary tape deck, or on two tracks of your master tape.

As a final incentive, Buff feels that a completely equipped console may actually be lower in price than a conventional board, since the cost of the automation is more than offset by elimination of countless controls. ■

The console's automation is cannily designed to give an engineer octopus hands and a stupendous memory.

JOHN BORWICK

Meet Neve's Necam

NECAM stands for Neve Computer-Assisted Mixing, representing a wholly novel approach to automation in the recording studio. It's built by Rupert Neve and Company, who, operating from a tucked-away model factory in the English countryside, have built up a formidable reputation for superbly designed and constructed mixing consoles which they have put into top studios in all corners of the world—80 per cent of their production is exported.

When Neve throws its particular hat into the automation ring, we can be sure it is with an entry to be reckoned with. Neve design engineers not only know how to make very good mixing desks, but they spend countless hours in working studios with the wizards of the knobs and tapes, becoming aware of the problems that crop up there daily.

The first thing that hits you between the eyes when you see NECAM in action or get the feet of it for yourself is that the fader slide controls actually move up and down continuously during playback to the positions they occupied at the time of the original mixdown. This is similar to what one sees in the control cabin of a modern aircraft when the auto-pilot is in use: all the knobs and levers continuously move to their operative settings as if manipulated by invisible hands. The reasons for this are the same as those that dictate the NECAM design—(a)to let the knobs themselves act as indicators and (b) to enable the engineer to take manual control instantaneously.

According to Geoffrey Watts, Neve's engineering manager, when designing NECAM they took five factors as their criteria, aiming to eliminate dull and repetitive tasks on the console, speed up mixing operations, and offer the engineer positive help towards enhancing his artistic results. The main idea was to give him more hands, effectively, and a greatly improved memory so that he could concentrate on optimizing the finished product.

Here are the five Neve desiderata: integrated tape control, instinctive update, unlimited recall, merge capability, and time independence.

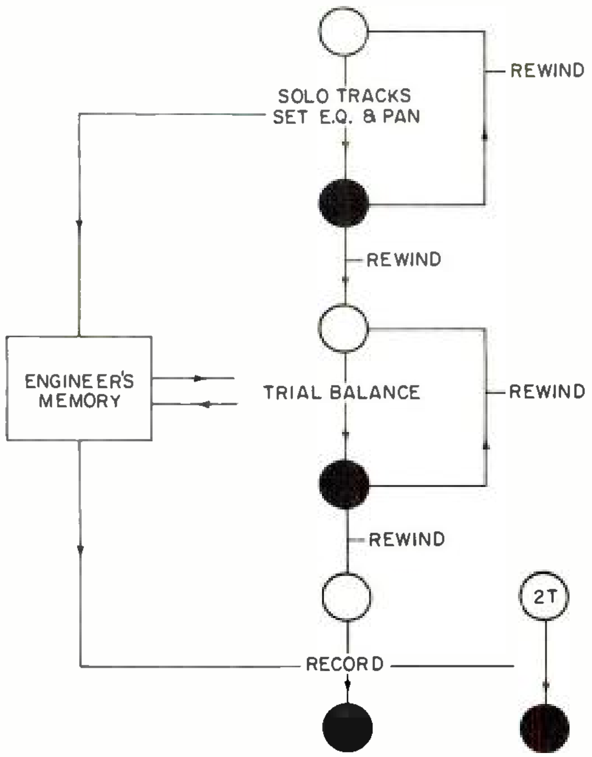
INTEGRATED TAPE CONTROL

The main situation for which NECAM was designed is where a multi-track master tape has been compiled—perhaps after many hours of overdubbing—and now has to be mixed down to a two-track stereo (or possibly four-track quadripronic) master for processing. A typical remix flow chart is shown in FIGURE 1. The multi-track master must be run and rewound several times—first to allow the engineer to set such treatments as e.q. (frequency correction), pan, and reverb; secondly, to try various level balances before the optimum mixdown is recorded and finally approved. Clearly this involves a great deal of rewinding to a cue, which may or not be the start of the track, putting a considerable strain on the engineer's ability to remember what he did, when, and where.

The NECAM solution is to harness a small computer to the sound mixing console (it is the Model LSI 2/10 produced by Computer Automation) and program it to fulfill all the engineer's locate—store—recall—control—merge functions at the touch of a button. A key feature is the laying of a data track on the multi-track tape itself (see FIGURE 2). This robs the studio engineer of one of his signal tracks, so that he must make do with 15 music tracks instead of 16, or 23 instead of 24, but it is actually simpler than some other proposed systems which either require two data tracks or the complication of a special synchronized auxiliary data recorder.

The data track is a simple audio pulse one comprising a time code to which the computer relates both memory and command functions. Any number of sequential mix-down runs can be stored in a separate memory bank, consisting of a "floppy disc" store, to be recalled later. The time

Figure 1. Remix Flow Chart.



John Borwick is Audio Editor of the British publication "The Gramophone" and is Senior Lecturer, Recording Techniques, in the Department of Music at the University of Surrey, England.

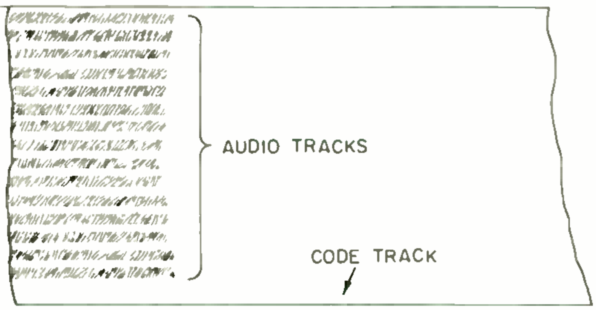


Figure 2. Typical Code: 07 38 46 09

code employed is the standard SMPTE edit one, with eight digits representing hours, minutes, seconds and "frames" so that interface with vtr and film systems is simple. Because it is tolerant of drop-outs, splices, and even leaders in the tape it can be assigned to an edge track. The computer actually scans five frames of the data track simultaneously to check their validity.

All the usual transport remotes are provided—wind, rewind (with the tape away from the heads), play and pause. The term *pause* is used rather than *stop* because, though the transport is halted, the take is not terminated until either the *keep* or *clear* button is pressed. All functions are programmed from a display-control unit which is shown in the photograph (FIGURE 3) and at the center of the schematic diagram (FIGURE 4). A most useful feature is that the mix can be tackled in segments of only a few bars at a time if necessary. These are identified by labels, which are simply numbers which the computer associates with the time code, and define the start and finish of any segment. Up to 999 separate points may be so defined—which should be enough for even the most complex album side or wayward producer.

The display-control unit, besides indicating the time code and mode in use, has a number of polite messages for the user such as *tape has reached "TO" Label* (end of a segment), *Please touch required faders* (when sub-grouping is desired), *Label X has not been defined* (when the operator has asked for a "wrong number") and *Storing labels—please wait* (the system has acquired more than 30 labels and is therefore quietly transferring them to the Data Disc).

INSTINCTIVE UPDATE

It goes without saying that any automated system must give the operator the ability to listen to a previous mix and then update it by altering the balance setting of his faders. In other approaches to this problem, the faders have been

Figure 3. The display control unit.

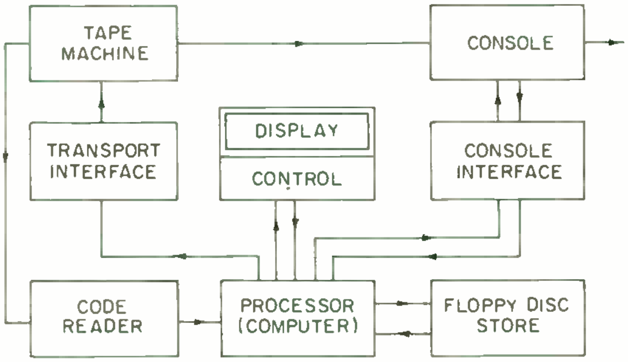


Figure 4. Key parts of the system.

converted into d.c. potentiometers whose voltage outputs have in turn controlled the gain of vca's (voltage controlled amplifiers). An analog-to-digital converter (adc) has been needed to translate the instantaneous voltage at the slider of each fader into digital terms, while the resultant information has been stored in real time on one track of the multi-track tape. In this way, every move of the operator during a trial mix can be stored alongside the music tracks as he operates.

On replay, a digital-to-analog converter (dac) can retrieve the control voltages and drive the vca's so as to repeat the mix exactly as the operator did it before, without his needing to touch a control. But when the engineer wants to perfect the mix by updating the stored information, there are problems. Clearly he cannot reconnect a given fader unless it is first set to its instantaneously correct position—otherwise a step in level would be introduced. Some systems, therefore, have a small meter by each fader, reading the dac output and perhaps supplemented by pairs of lamps.

Others make the vca level dependent on the sum of the dac and fader outputs, so that the operator has all his faders set to zero during the first replay and makes amendments by raising them. But there are drawbacks to this arrangement, in that he may unwittingly exceed the permitted input swing to the vca and of course he has lost

Figure 5. Automated control of the faders.

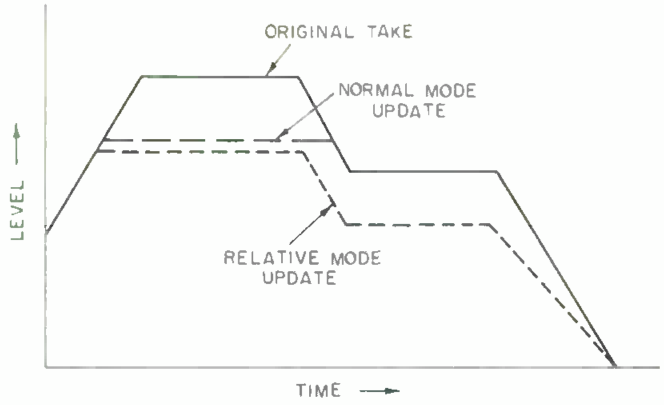


Figure 6. Any correction made manually will apply proportionately to all remembered movements thereafter.

the unique indicating feature whereby the knob positions give a graphical display of the levels within the mix.

Worse still, both the above systems take up a second track on the tape for the amended mix, and they can only remember one previous mix at a time.

The NECAM system can store literally dozens of mixes and yet it only steals one track of the multi-track master. The servo-motor driven faders give continuous indication of the moment-by-moment level on each channel or group of channels, as already mentioned. The faders are of the normal plastic track type and control audio levels in the usual way, thus avoiding the limitations of vca circuits. The fader knobs are touch sensitive (that is they incorporate a proximity switch which tells the computer to record the changed information) and the operator simply follows his instinct during a re-run. He can change from the *read* to the *write* mode by resting a finger on the faders he wishes to rebalance on the new take.

FIGURE 5 is a close-up photograph of the NECAM faders. A three-position switch can be seen above each one. This provides versatility in update in three distinct modes:

(a) *Manual mode* disables the fader servo, to cancel any remembered activity and allow the operator to choose settings at will—which will, of course, be stored this time round.

(b) *Normal mode* permits direct control by the computer, so that when the operator releases contact with the fader knob after making an adjustment, the fader remains stationary until the next stored moment arrives, when it reverts to that stored position. This is shown in the level/time diagram of FIGURE 6.

(c) *Relative mode* decrees that any correction made manually will apply proportionately to all remembered movements thereafter (see FIGURE 6 again). This allows one track to be lifted or decreased for the rest of the whole mix without the need to "ride the fader."

Figure 7. A view of the system in action.

UNLIMITED RECALL

As has been mentioned, the system can commit to memory store dozens of trial mixes and recall any of these at a moment's notice. When the final mix data, plus any others required, are stored in the floppy disc, which resembles a thin 45 rpm record in its sleeve, the disc fits neatly in the tape box along with the tape.

MERGE CAPABILITY

The advantages of the NECAM method of relating all functions to the time code track really come into their own in the *merge* situations. Without the need even to move the tape, the computer can be requested to build up a final take—just by handling data—which can comprise short segments of the two-track copy made during different mix attempts (this corresponds to electronic butt splicing and, since each segment will normally follow the ending of the prior take, no discontinuities in the music signal will exist), or even assorted subdivisions of individual tracks. For each merge trick, the operator gives simple routine instructions to the keyboard and the computer does the rest.

TIME INDEPENDENCE

Mixdown sessions can be long and exhausting, with vital artistic and technical decisions often made in the small hours of the morning. A computer assisted system like NECAM—which has a perfect memory, will recycle short or long takes, butt or overlap them, even work at half speed and store as many trial mixes as necessary—will encourage studio personnel to break off and resume work when they are fresh. Since the floppy disc store can be packed with the multi-track master, the whole history of the remix sessions can be preserved.

Notice too that there is an advantage not so far mentioned. With the appropriate tape machine interface at the disc-cutting channel, it would be possible to cut straight to disc from the multi-track tape without the need ever to have made a stereo or quadriphonic master tape—eliminating a generation of quality-degrading tape copying.

TO SUM UP

Neve's NECAM has clearly got a lot going for it and studio managers will be seeking more detailed information and demonstrations. As I write, you need to visit England to see all the hardware and software in action. But I am told that Geoffrey Watts is considering making a video recording of his demonstration and it might be possible to view this in the USA or anywhere else on request. A view of the NECAM system in action is shown in FIGURE 7.

Some may argue that this return to electro-mechanical devices such as servo-motors is a retrograde step, since these have a past history of poor reliability. However, Neve argues that today's devices have high reliability; no clutch system is used, so manual operation of the fader is against the load of the motor and just gives a smooth damped feel.

It might also be asked why the computer-assisted operation in NECAM has been applied only to channel and group faders (plus perhaps such obvious extras as master reverb send and return). Neve says that extending this to cover e.q., stereo width, panning, etc. would present considerable space and cost problems. Their discussions with numerous studio engineers have highlighted the mixdown balance situation as top priority for automated control at this time. Naturally, inasmuch as every Neve control console is a one-off custom job for a particular client, they are ready to design a NECAM installation with just as few or as many control functions as requested. ■

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(continued)

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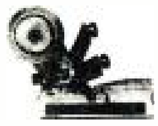

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db people/places/happenings

• **Switchcraft, Inc.** of Chicago has purchased the thumbwheel and rotary switch lines of **Chicago Dynamic Industries, Inc.** The new line has been incorporated into Switchcraft's production at their Chicago facility.

• Management and marketing structures for the Cetec Broadcast Group formed of five divisions of the **Cetec Corporation** have been finalized. **Robert M. Ward** will be responsible for group sales and marketing with headquarters in Satna Barbara, Ca. **Jay Cook** has been named advertising manager. The five divisions are Schafer, which produces automated radio systems; Jampro Antenna, f.m. and t.v. transmitting antennas; Sparta, transmitters and radio studio broadcast systems; Vega, wireless microphones, and Cetec Audio, studio consoles, professional tape reproducers, and loudspeakers.

• **Allen and Heath, Ltd.,** manufacturers of mixing equipment, and tape deck manufacturer **Brenell, Ltd.** have joined forces. Their combined facility is located at Pembroke House, Campsbourne Rd., Hornsey, London, N8 England.

• **Joachim P. Diermann** has been appointed chief engineer of the audio-video systems division of the **Ampex Corporation**, Redwood City, Ca. Mr. Diermann has conducted development work at Ampex, with whom he has been associated since 1966. Among his projects were electron beam recording, computer and data technology and the Electronic Still Storage system.

• **Sigma Instruments, Inc.** of Braintree, Mass, has appointed **Jerome J. Tuite** as regional sales manager for switching products. Mr. Tuite will be responsible for sales and marketing in the eastern region. Before coming to Sigma, Mr. Tuite was with the **Siemens Corporation.**

GEORG NEUMANN

Georg Neumann, a pioneer in the field of audio recording and the inventor of the gas-tight nickel cadmium battery, died at his home in West Berlin on August 30, 1976 at the age of 77.

Georg Neumann was born on October 13, 1898 and has been closely associated with the development of sound recording and reproduction techniques since the middle twenties. As a young engineer, he developed a carbon microphone, known as the Reisz microphone, displaying remarkable improvement over carbon granule types then in common use. In 1928, Georg Neumann produced the first commercial condenser microphones and since then these have been preeminent in their field. The high fidelity recording was born with the advent in 1948 of the long playing record recorded with the now-classic Neumann U47 condenser microphone, and Hi-Fi became a household word as a result.

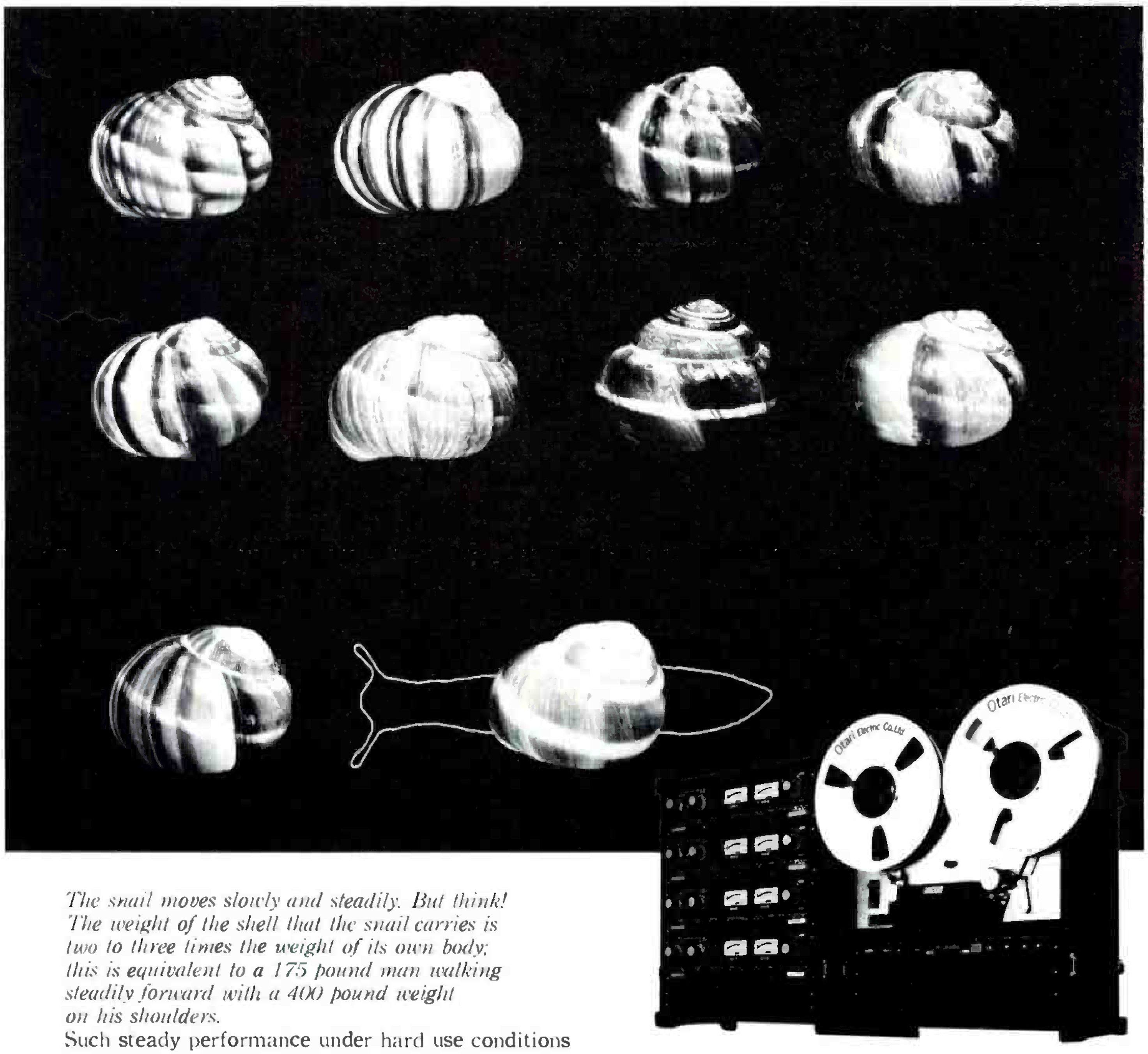
In 1930 Georg Neumann developed the first of a long line of disc mastering lathes, as well as one of the first electro-mechanical disc cutting heads and in 1957 his company introduced the world's first stereophonic disc recording system.

Some of his other inventions covered the first linear motion pen level recorder, a device used by all manufacturers of such recorders today. During World War II, Mr. Neumann developed, and soon after the war, patented, the world's first gas-tight rechargeable nickel cadmium battery, without which space exploration would not be possible.

Georg Neumann was awarded Honorary Membership in the Audio Engineering Society in 1973. In March, 1976 he was awarded the Society's highest honor, its Gold Medal, which he accepted in person at the Society's European Convention held in Switzerland.

He is survived by his wife Elly, and two children, Dr. Ralf Neumann and Mrs. Ingrid Canetti, both living in Paris.

once is enough!



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