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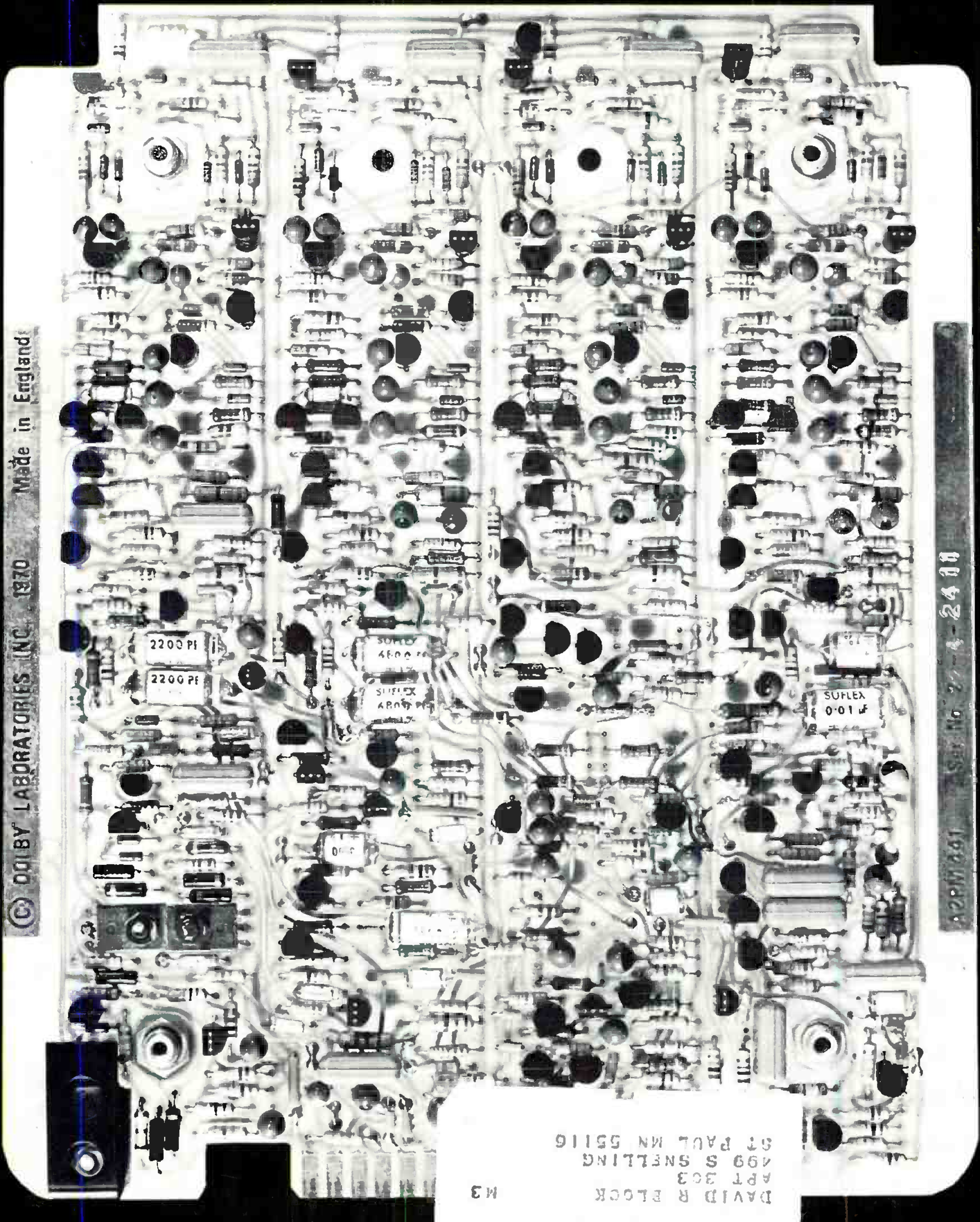
THE SOUND ENGINEERING MAGAZINE

FEBRUARY 1972 $1.00



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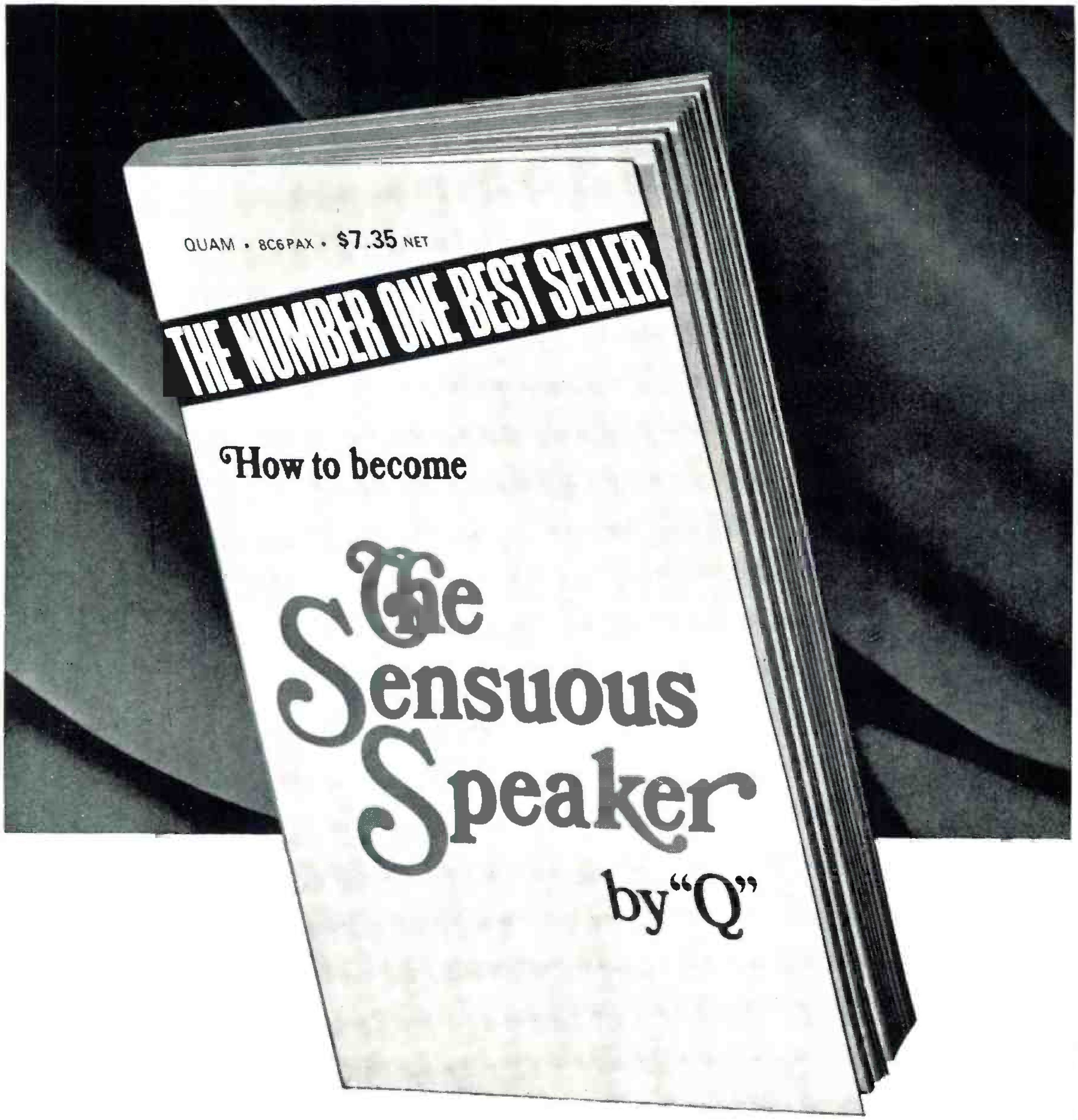

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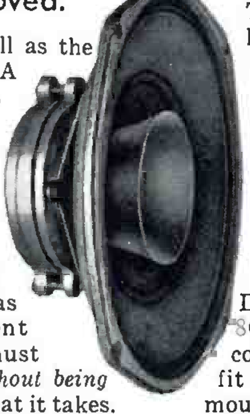

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COMING NEXT MONTH

• THE TAMING OF MAXWELL HALL by William Mattheus. This article describes the sound system that had to be installed in a difficult hall to accommodate Billy Grahame's Crusade when it came to town. The lessons in sound-reinforcement technique will be of use to all.

William Kaffenburger of Evanston High in Illinois was faced with a unique miking problem for a performance of Ahmal and the Night Visitors. The solutions were based on unconventional approaches to the problems.

Last month space crowded out Robert Hawkins RADIO PREMIUMS story. This is a delightful look at the golden age of radio giveaways. A bit of nostalgia for everyone who has toiled in broadcasting for a while.

db VISITS. This time our peripatetic camera crossed the border into Canada and went to the opening ceremonies of Manta Studios. Manta is Toronto's most modern with ample studio space and interesting layouts of sophisticated equipment.

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

ABOUT THE COVER

• There is beauty in electronic circuitry. The board on our cover is the conurbation circuit board from a Dolby 360-series unit. No less than 500 components resulting in 1000 solder joints are on it. The photo is from Dolby Labs.

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THE SOUND ENGINEERING MAGAZINE

FEBRUARY 1972 VOLUME 6, NUMBER 2

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letters

The Editor:
Per the discussion of room equalization in September:

"If the system, including the air path from the loudspeaker to one position in the auditorium, is made flat, it will not, in general, be flat for other positions or for other paths in the room."

The above was published in 1934. Not 1954 or 1964 or 1971 but 1934. Symposium on Auditory Perspective, Staff of the Bell Telephone Laboratories, Trans AIEE (Electrical Engineering, Vo. 53, No. 1, pp. 9-32, 214-219. The specific quotation is from System Adaptation, E. H. Bedell and Iden Kerney; see page 218, column 2.

Room equalization aims to render the sound reinforcement system in one enclosure, room hall, or auditorium the same as in any other enclosure, similar or dissimilar. Somehow this seems to be like asking the Boston Symphony in Symphony Hall to sound the same as the New York Philharmonic in Lincoln Center for the Performing Arts.

No doubt the acoustics of Symphony Hall could be modified electronically so recordings made there would be like those made in Lincoln Center. Would that be good?

Paul Klipsch
Hope, Arkansas

advertisers index

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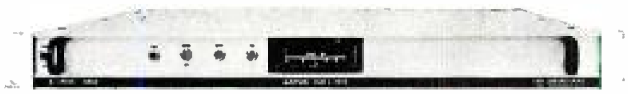

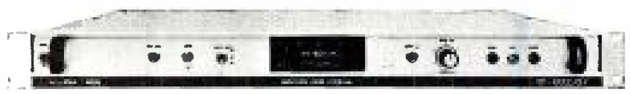

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George Alexandrovich

THE AUDIO ENGINEER'S HANDBOOK

Troubleshooting techniques revived

● Topics for my monthly columns usually come from my daily experiences and encounters with different problems. Like most audio pros I can't stop thinking about the projects I am working on during the day when I get home, and consequently my columns reflect that. When I experience an epidemic of calls from the younger group of audio engineers, technicians, and maintenance men, I get an urge to put my answers in writing to benefit some of my friends who did not yet call me.

So, the topic for this issue is troubleshooting. Most troubles are concentrated around electronic parts of the system where you don't see them to squirt oil, tighten nuts, or adjust tension. We usually find out about them when it is too late; then we either start hearing them, smelling them, or learn about them from customers over the phone. And this is how each of us slowly develops his own ways to combat things such as oscillations, motorboating, hum, pops and clicks, crosstalk, distortion, wrong gain, poor frequency response —you name it, we have to fight it.

Strangely enough, just as wrestlers in the ring, a person develops certain methods in troubleshooting—a favorite hold. After a while you know if your method is a success and brings victories. You change your approaches as you go along until you reach the state when a defect or malfunction can be located and rectified in a shortest possible time.

Many distinguished engineers and scientists have written on the subject. My reason for daring to write again is only because I believe that every person adds a little of his own specific methods—which may be better suited for the technology of today. My favorite approach to troubleshooting consists of the following; get all the facts you can; start activating the system slowly, working with each input and each function separately; once trouble is detected use a further process of elimination to pinpoint the fault and eliminate it.

In practice, you are often faced with a situation when symptoms are described vaguely and incompletely. It is my suggestion that before you decide to repair the trouble, double check the symptoms yourself. You won't be sorry.

Consider that you are responsible for the operation and maintenance of a recording studio. One day an engineer comes running to you for help, saying that he has a noisy recording. What should you do—roll out the spare tape machine, or check to see if something else is causing the trouble. But whatever you do, don't allow yourself to be influenced by what may be somebody else's erroneous conclusion. Confirm this to yourself by checking the system again. Try to extract as much information as possible from the engineer to save time. Find out if monitor sound is clean when you monitor the console and not the tape. Is hum or noise still present when the tape is stopped; is the noise

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recorded or does it appear only when the machine is in a playback mode?

What I am trying to say is that one has to act as a Sherlock Holmes, gathering all the facts about the malfunction, even if you have to start checking all the circuits yourself in order to complete the picture.

It is impossible to easily teach somebody how to troubleshoot effectively, but one can be helped if he is told what to look for. I am going to divide symptoms into separate paragraphs and list all possible causes that come to mind.

Hum. Possible causes are: disconnected shield, defective power supply, audio transformer proximity to the low level wire to the power line. In mechanical systems, the cause may be the rumble from the turntable motor appearing as a hum, or vibration of the building registered as a rumble and taken for hum. Air-conditioning noise or pulsation can affect the microphones and can be mistaken for hum.

Cures for hum problems consist of repairing the power supply, moving of system components, grounding of the shield, shock-mounting the turntable. When microphones pick up air-conditioning noises, the use of directional mics or installation of dampers in the ducts will alleviate the problem.

Motorboating. This problem is generally associated with the power supply problems, especially when high-gain class AB amplifiers are in the system. Insufficient decoupling or high source impedance power supply is usually the cause. Less frequent cause is high-frequency oscillations producing instability at low frequencies.

Cures are; increased decoupling between the parts of the system by increasing decoupling capacitors, check output impedance of the power supply by measuring the voltage drop when a maximum load is connected — and then knowing the load impedance calculate using Ohms law. Source impedance of a few ohms generally means trouble. Typical readings should be in fraction of an ohm. Adding capacitance across the power bus may help. With the power supply having remote sensing—use it. This will shift the lower impedance point closer to the load. Also, check phasing of the signal at the low end as well as for the low-frequency response, if it extends below the useful range.

Frying noise. Frying noises are usually an indication that one of the semiconductors is defective or some resistor is overheating from the excessive current through it. Also partial demodulation of an r.f. signal may be mistaken for frying noise.

In a system, cure for frying noise should consist of isolating and replacing the defective amplifier or component. In an amplifier the cause of the failing resistor or transistor has to be found before the unit can be put back into operation, since the unit will fail again (possibly with more disasterous results causing more damage). Just replacing defective part without finding the cause won't do.

Pops and clicks. We should distinguish between pops and clicks detected in the system when no switching is done (susceptibility of the system to pickup transients originated by the equipment in the vicinity or in the same building) and the clicks produced when switching is done in the system. In the first case the cause (and consequent cure) are almost the same as for the hum problem—improper shielding or grounding, insufficient decoupling or poor filtering in the power supply, as well as unshielded and extra long sensing wires from the power supply to the load.

Cure in this case also consists of adding line filters if all the above-mentioned faults are corrected or brought under control without improving the condition. One of the best generators for testing the system for susceptibility to pops and clicks is usually an electric shaver or electric drill. They produce more transients than the vacuum cleaner or elevator do.

If the pops and clicks are a function of switching, the following could be the cause for it: no noise suppression of the relay coils, tape start or stop switches have no arc suppression capacitors across the contacts, in switching audio there is d.c. on the switch, change of termination value in the directly-coupled amplifier. In a correctly-designed system there should be no reason for expecting clicks or pops, but sometimes electrolytics become leaky and d.c. get on the switch contacts. Or additions and modifications of the circuit are made without incorporating noise-suppression components.

Oscillations. Causes: high gain, loss of grounding or shielding, phase reversal, loss of power supply regulation, or proximity of low-level lines to the high-level lines or conductors.

Cures are more or less self explanatory. When testing for phase shift or reversal, connect a diode across the terminals of the signal generator, then note which peak of the sine wave is clipped, upper or lower. Follow with a scope through the system looking for the identical scope picture. If the clipped portion of the sine wave shifts from the bottom to the top or *vice versa* you have phase reversal. Before changing anything check the diagram because one phase reversal may be of necessity followed and corrected by another. Today, when most of the mixing is active (that is, operational amplifiers, are being used as summing amplifiers), phase reversal is to be expected. This reversal is usually followed by another reversal in the output transformer.

Crosstalk. Causes range from improperly aligned tape recorder heads, a defective power supply, poor amplifier decoupling, defective summing amplifiers, to missing terminations.

Cure consists of using the process of elimination—working initially with two channels, feeding signal first into one channel at operating level and measuring the output in the other channel. While measuring crosstalk, manipulate faders of both channels to determine which part of the circuit initiates crosstalk and which is picking it up. Check, using the scope, for audio signal in the power line. This may indicate if the trouble is in the power bus. In a tape machine, before you start realigning heads make sure that all your assumptions that the fault is there are correct. Check patchbays, you may have something patched in that causes crosstalk.

Distortion. Possible causes include low supply voltage, voltage imbalance—especially in bipolar supplies, improper distribution of gain (levels), high-frequency oscillations, improper bias on the tape machine, faulty amplifiers.

Except for the distribution of gain, all cures are part of the standard troubleshooting routine. If the schematic doesn't specify the gain settings, the following should be a good rule of the thumb: Set the operating levels as close as possible to 10-15 dB below the clipping point of the stage. This will assure you good noise, low distortion, and crosstalk.

Low output. Sometimes, faults which show no obvious malfunctions as far as distortion or noise are concerned can be harder to find than completely dead circuits.

Causes for low output can be partially shorted or open circuits, improper patching or termination, and (believe it or not) high humidity. I have noticed that in in some operational amplifiers gain can be affected by the leakage across the circuit board terminals. Especially if f.e.t.'s are in the circuit.

Cure of this leakage consists of drying the boards and then covering them with a thin coating of a clear lacquer spray such as Krylon. Stay clear of contacts and controls. In order to determine which part of the circuit is at fault, follow the same procedures as suggested before—test the levels at each stage and compare them with the prescribed values. Make sure when you do this, that all compressors, limiters, and equalizers are set in the *off* position. Quite often you may have an equalizer set at −10 dB position at low frequencies and it just happens that the generator is also set for 100 Hz instead of 1000 Hz. You then assume that you have a low gain problem.

Poor frequency response. This is one of the most familiar problems of almost every type of system. It can be the result of an impedance mismatch, a shorted or open circuit, improper level setting, head alignment in the tape machine, extra long shielded wires or a poorly oriented knob on the equalizer.

As far as impedance mismatch is concerned, some circuits require proper termination or they may exhibit either high-frequency boost with insufficient loading (transformers), or high-frequency droop with excessive loading. Passive equalizers are also impedance sensitive. Long shielded wires from a magnetic cartridge will roll off high-frequency response (don't use more than 100 pF of cable capacitance for a 50k ohm cartridge). Insufficient low-frequency response can be the after effect of a failing electrolytic on the input with consequent insufficient capacitance.

What I have just written is only a small part of what can be said about the problems of troubleshooting. You can not miss the constant appearance of magazine articles discussing specialized troubleshooting. I think that we should attempt to do the same for the audio field. If those of you with problems and questions will write to me, we can discuss them openly for the benefit of all. Naturally your name and affiliation will not be revealed unless you wish it. Send me sketches, drawings, notes, curves—anything that may pertain to your problem and will help me give you advise or a possible solution to your problem. Don't expect me to solve all problems from my desk, but I will do my best to be of use to those who need my help. You will help yourself as well as others. ■

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• The annual **Midwest Acoustic Conference** will be held Saturday, April 15, 1972 with the broad theme "Four Channel Sound Reproduction, Creation and Re-Creation of a Sound Field." The all-day meeting will be held in the Auditorium of the Technological Institute of Northwestern University in Evanston, Illinois. Professionals and non-professionals interested in sound reproduction are invited.

The conference is held annually in the Chicago area under the auspices of the Chicago regional chapters of the **Acoustical Society of America,** the **Audio Engineering Society;** the **Audio and Electro Acoustic Group** of the **IEEE** and the **Chicago Audio and Acoustics Group.**

The tentative program outline calls for a 9 a.m. start under the chairmanship of **Daniel Queen,** acoustics consultant and head of Daniel Queen Associates.

A discussion on **What is Ambiance?** is scheduled for 9:30 a.m., under the chairmanship of **Marvin Camras,** senior scientific advisor, I.I.T. Research Institute. **John Volkman** is scheduled on the panel.

Bruno Staffen, applied acoustics consultant for the consumer products division of Motorola, Inc. will be chairman of the session convening at 11 a.m. Subjects for this meeting will include Reproduction in Rooms and Physical and Psychoacoustical Considerations. **Roy Allison** of Acoustic Research and **Dr. Amar Bose** (Bose Corp.) are panelists.

Following lunch, **James Cunningham,** Eight Track Recording Corp. of Chicago will chair a 1:30 p.m. session on Recording for Ambiance and Effect. **John Eargle** and **Bill Putnam** are panelists.

James Kogan, vice president and director of research, Shure Bros., Evanston, Illinois, will lead a 3:15 p.m. meeting on Recorded Media and Discrete vs. Matrix Format. **Duane Cooper, Warren Eisem,** and **Howard Durbin** are panelists.

Demonstrations of four-channel sound reproduction will take place at 7 p.m. following a dinner at 5:30 p.m.

It should be noted that the panelists listed are from a tentative list that will be augmented. Information on the conference may be obtained from the president of the Midwest Acoustic Conference's Executive Committee, Mr. Daniel Queen. He may be reached at **Daniel Queen Associates, 5524 West Gladys Avenue, Chicago, Illinois 60644.** His telephone is **(312) 261-5738.**

Norman H. Crowhurst

THEORY AND PRACTICE

● To pursue the problems of controlling bias in transistor circuits, consider the circuit shown in FIGURE 1. The 20-ohms load is a nominal representation of the base input resistance of a next stage, which is *not* a constant resistance. But giving it a value of 20 ohms enables us to do some figuring.

Assume the beta of each transistor is 100. Using the collector to base bias, this makes the second transistor look like a d.c. resistance of 2.2k divided by 100, or 22 ohms. In series with the collector resistor of 100 ohms, the 12 volt supply divides approximately into 10 volts across the 100 ohms and 2 volts across the transistor. The transistor current is determined by 10 volts across 100 ohms, at 100 milliamps.

The base bias current is about 1 milliamp. When feeding into a 20-ohm load, increasing base current to 2 milliamps will make collector current 200 milliamps, of which 120 milliamps flow through the 100 ohm resistor and 80 milliamps through the 20-ohm resistor. Going up to 2.2 milliamps will saturate at 220 milliamps, with 120 milliamps through the 100 ohm and 100 milliamps through the 20 ohm.

In the reverse direction, when base current is cut off, all the 100 milliamps through the 100 ohms is momentarily shunted through the 20 ohms as well. The coupling capacitor averages out the a.c., so that the flow of 100 milliamps each way through the 20 ohms maintains the balance. Taking a quick look at the first stage, if its beta is 100, the transistor will look like a d.c. resistance of 220k divided by 100, or 2.2k. This, in series with the collector resistor of 4.7 puts 3.8 volts at the collector, with about 1.75 milliamps.

This should enable this stage to deliver ±1.75 milliamps to the base of the second stage, without difficulty. As it only needs ±1 milliamp, or maybe slightly more, so far so good. But let us consider three possibilities of variation. (1) if beta changes from 100 to 70 or 140; (2) if the output load of 20 ohms changes; (3) if the input drive reaches the full 1.75 milliamps available from the first stage.

If beta of the second stage changes from 100 to 70 or 140, the d.c. resistance of the transistor changes from 22 ohms to about 32 ohms or about 16 ohms respectively, so that collector voltage changes from around 2, to around 3 or 1.5 volts respectively. The stage still works, although maybe not at optimum to deliver maximum current into the 20-ohm load.

If the 20-ohm resistance drops below 20 ohms, the voltage developed across it will drop, but the current swing will not be changed materially. The improper things happen if the 20 ohms runs high. Suppose it comes off

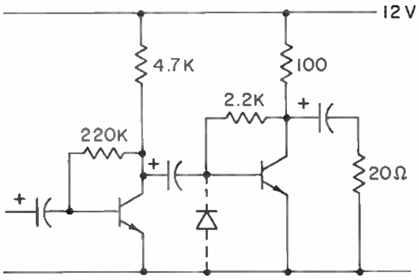


Figure 1. This is a circuit to consider.

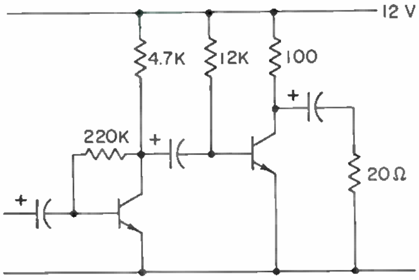


Figure 2. Here are other ways to get the bias.

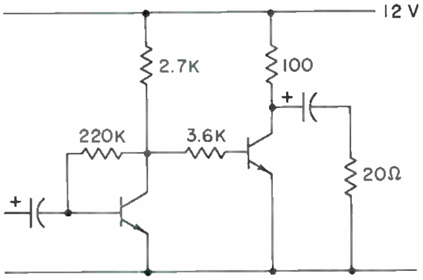


Figure 3. This circuit has the advantage of being direct coupled.

altogether. Now, when the current drive is negative 1 milliamp, to cut-off, the collector runs up to +12 instead of about +4 (maximum with load connected). On the drive in the other direction, the transistor will saturate at 120 milliamps collector current, instead of about 200 milliamps with the load connected. This means it will saturate at 1.2 milliamps (only 0.2 milliamps a.c. component) in that direction.

So, over the cycle, the average collector would be about 3.5 volts, instead of 2 volts, and the 2.2k bias resistor will increase collector current, until it is saturated for more of the time, only amplifying part of the positive peaks.

The third possibility is that the drive reaches ±1.75 milliamps, because that is what the first stage will deliver. In this circumstance, the top 0.75 milliamps in the negative-going direction will cut off the output transistor, whatever its load, because its bias current is only 1 milliamp.

Current beyond the saturation point, in the other direction, also acts purely as rectified a.c. piling up negative charge on the output side of the interstage coupling capacitor. This has the effect of increasing the amount that the first direction runs into cut-off. Again, only part of the waveform gets amplified, but for a different reason.

So let us look at some other ways of getting the bias. FIGURE 2 uses a 12k from supply positive, instead of a 2.2k from collector. When beta is precisely 100, this will yield the same operating point for normal operation. For the second deviation, of removing the 20-ohm load, it will prevent the bias current from changing, because bias current will be held steadily at 1 milliamp (by 12k across 12 volts), come hell or high water.

But changing the beta will change the collect current from 100 milliamps to 70 or 140 milliamps. Actually 120 milliamps is saturation, with the 100-ohm collector resistor, so the stage ceases to work when beta reaches 120. At beta of 70, the collector voltage rises from 2 volts to 5 volts, considerably reducing the available output swing into the 20-ohm load.

We hardly need consider the effect of increasing the drive from 1 milliamp to 1.75 milliamps a.c. peak. The output stage runs into saturation faster, because the self-correcting effect of the collector feedback is missing.

There is another factor: the effect on the dissipation of the second transistor. The first circuit sets the operating point at 2 volts 100 milliamps, which is a dissipation of 200 milliwatts. This would be safe for a transistor rated at 300 milliwatts maximum, although when the collector voltage rises to 3 volts, the dissipation is 270 milliwatts, which is getting close.

However, with the biasing circuit of FIGURE 2, if beta drops to 70, collector voltage is 5, current 70 milliamps, a dissipation of 350 milliwatts. The margin of safety has more than gone.

So let us look at the circuit of FIGURE 3. This has the advantage of being direct coupled, so that increasing the drive beyond ±1 milliamp will not cause overbiasing of the stage. It will merely over-drive the second stage on an instant-to-instant basis.

Assuming a collector voltage of 3.8 volts for the first transistor, the 3.6k coupling resistor passes about 1 milliamp, which also passes through the collector resistor. This raises the current through the latter from 1.75 milliamps to 2.75 milliamps, requiring its value to be dropped from 4.7k to 2.7k to retain the same voltage drop. As the transistor still operates at the same collector current, the 220k bias resistor is unchanged in value.

Now, if the 20-ohm resistor changes in value, this does not change the bias of the output stage, although change in the beta of the first stage can change the bias of the second stage, by changing first stage collector voltage, at the "front end" of the 3.6k resistor.

And what about change in beta of the second transistor. Just like FIGURE 2, assuming that collector voltage of the first stage is fixed at 3.8 volts, bias current is fixed by the 3.6k resistor, as it was by the 12k resistor in FIGURE 2.

So, all in all, the circuit of FIGURE 1 seems best, except that the circuit of FIGURE 3 eliminates the time-constant of the interstage coupling capacitor. This can to some extent be remedied by using a suitable base to ground resistor. In some circuits it may have more benefit than here.

Assume the base to emitter voltage is 300 millivolts, with +100, −300, for the ±1 milliamp swing. This looks like from 100 to 300 ohms base input resistance. At cut-off, this suddenly rises almost to infinity, so that the only resistance to bring the circuit back to normal is the 2.2k from the collector. This is certainly better than the 12k of FIGURE 2.

Another trick that helps, by utilizing the drive beyond cut-off to discharge the rectified effect on the coupling capacitor, is to connect a diode across the base-emitter junction of the transistor, shown dotted in FIGURE 1.

Possibly the best alternative, is to design the previous stage so that it is incapable of delivering a 75 per cent overload drive current to this stage. If you raise the 4.7k collector resistor to 8.2k, with appropriate change in collector to base bias resistor, so collector current is about 1.2 milliamps, instead of 1.75 milliamps, with collector at about 2 volts, requiring 150k or 180k, the previous stage will only be able to deliver about 20 per cent overload current, which will have far less detrimental effect. ■

attention overseas readers

The slow delivery of each month's issue to overseas addresses often results in the return of reader service cards that have passed the date of their expiry. As the processing computer is trained to reject such cards, overseas readers wanting information are not always receiving it.

You should check the expiry date on the reader service card. If the date is close, and you must allow the time for mail service back to the address on the card, we can only urge that air mail be used.

John M. Woram

THE SYNC TRACK

• The letters continue to arrive at **db Magazine,** and at studios throughout the country, if the mail at Vanguard is any indication. "How do I become a recording engineer?", they ask. As many as possible get answered, but to respond to them all would be an almost full time job.

And, what does one say by way of answering? Becoming a doctor or a lawyer is certainly easier (to describe, that is). Just go to school, get your diploma and open up an office. Next question.

The would-be recording engineer's path is not so clear, though it may be just as long. There are still no universities offering courses leading to a degree in recording-studio engineering, which may be a very good thing, come to think of it.

To digress; my bride has a deep interest in medicine. Now that the kiddies are almost as big as their parents, she has been working at the nearby nursing home. She gets some satisfaction helping the aged through their endless last days. And she goes to the State University at Farmingdale in the evening to take courses leading to an Associate in Applied Science degree in Nursing. She just passed her final in English Composition, which means she'll now be able to take Introduction to Literature when its offered. She'll be a fine nurse, providing she doesn't flunk Speech I and does well in her Sociology course. In the meantime, the patients can just wait a bit longer for better care. You certainly wouldn't want *your* parents being touched by someone who couldn't tell a verb from a preposition would you?

Now, being an unlettered type myself, (*hardly, ed.*) I would say, if I were asked, that any educational authority who insisted that an otherwise well-qualified person be prevented from functioning as a nurse until she got through Introduction to Literature *et al,* was either kidding, or criminally insane. If this same mentality had anything to do with the recording industry, can you imagine the drivel one would have to wade through in order to get a degree in recording?

There is certainly nothing wrong with a good liberal-arts education (I wish I had one). If you have the time, money, and inclination it is well worth the effort. But, if you want to make a living in the recording industry, you don't have to sweat English Composition.

Perhaps the industry leans too far the other way. At times, there seems to be an actual suspicion of people with a degree. I'm not sure why this should be. Even if a degree does you little good, it certainly does you no harm. Unless it elevates you above the dirty work that is so much a function of the recording world. Even if you *really* don't mind "shlepping" mic-

Studio Technology and Practice

SURVEY OF RECORDING FUNDAMENTALS

A. Sound generation and propagation, simple harmonic motion, sine waves and music sources.
B. Hearing, dynamic range, loudness, masking, decibels.
C. Electrical fundamentals, ac circuits, RLC impedance.
D. Magnetic fundamentals, dynamic devices—mics., speakers, pickup cartridges, cutting heads, tape heads.

MAGNETIC TAPE RECORDING

A. Basic tape recording process, block diagram of tape recorder, introduction to high frequency bias.
B. Bias and erase adjustment, head alignment.
C. Multi-track recording, remix consoles.
D. Complete tape recorder alignment procedure.

STEREO DISK RECORDING

A. Disk groove geometry, lateral & vertical modulation, mono & stereo grooves, groove amplitude & velocity.
B. Disk playback systems & standards, velocity & level, test records, magnetic & dynamic cutting systems.
C. Stereo/mono compatibility, automatic pitch and depth control, master, mother & stamper plating, pressing.
D. Test equipment-oscillators, voltmeters, oscilloscopes, intermodulation & harmonic distortion meters.

MICROPHONES AND STUDIO CONSOLES

A. Lines-balanced & unbalanced, introduction to impedance matching, European & American systems.
B. Impedance matching, jack fields & patching, basic console layout.
C. Microphones and preamplifiers, noise considerations.
D. Limiters, equalizers and noise reduction equipment.

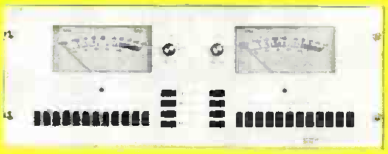
rophone cables, going for coffee, and cleaning up after the recording staff finishes, *they* may find it difficult to accept. This is certainly not fair to the degree holder, but I'm afraid many studios have gotten stung by overqualified help from time to time.

There is no question that classroom work could play an important part in the development of recording skills. In fact, several schools now offer an occasional seminar for those interested in learning more about recording. Yet, with the exception of the Institute of Audio Research here in New York City, few, if any, offer a continuing program of any considerable length.

The Institute of Audio Research was begun several years ago, in an attempt to provide classroom training directly related to the recording industry's needs. I've printed a typical working outline for one of the Institute's courses. Note the glaring absence of Introduction to Literature! With the exception of a few of the earlier sessions on acoustic and electrical fundamentals, most of the classroom time is directly related to recording. And by a remarkable coincidence, I'm going to be meeting with the class during the present term in an effort to put classroom theory into practice during a few lab sessions in the studio. Later in the term, after some first-hand experience with the Institute's program, I'll be in a better position to report on its contribution to the recording industry, as I reinforce my somewhat shaky background in theory, in exchange for whatever practical assistance I can contibute. ■

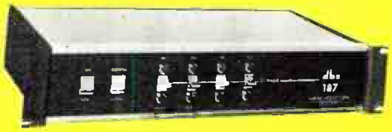
NEW PRODUCTS AND SERVICES

METER PANEL



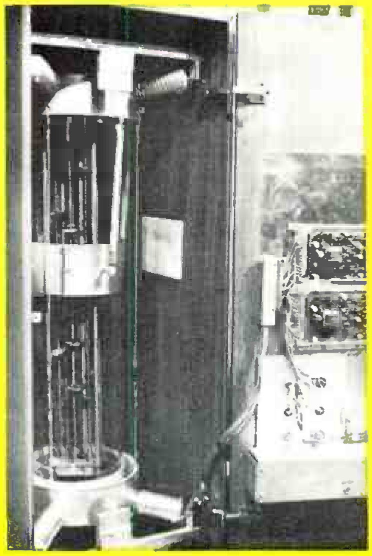
• A two-channel vu meter panel and attenuator is offered in this model VP-1S. The unit is designed to monitor any two of twenty-four inputs simultaneously. Standard input level is +4 dBm to +45 dBm dependent upon attenuation selection. Optional input preamps are available for levels down to 90 dBm (0 dBm). Attenuation selection is by four selection push button and a 4 dB vernier control. The unit fits a standard 19-inch rack.
Mfr: Broadcast Automation Associates
Price: $320
Circle 60 on Reader Service Card

NOISE REDUCTION SYSTEM



• Over 30 dB of noise reduction is claimed for the DBX 187, a four-channel noise-reduction system that decreases hum, hiss, and crosstalk in recordings. Straight-line decibel compression before recording and complimentary expansion in playback provides perfect transient tracking with no pilot tone or critical level matching. Hiss and asperity noise at normal levels are reduced by 10 dB. Four channels of record or play are provided.
Mfr: *DBX Inc.*
Price: $1950
Circle 62 on Reader Service Card.

REVERB SYSTEM



• Model BX-20E produces reverberation by the torsional vibration of a specially treated coil spring. The torsion elements consist of several units with microscopic and macroscopic variations of the line paramerer interspersed with controlled damping and supporting elements. Electronic damping utilizing motional feedback, permits variation of the decay time from 2 to 4.5 seconds. The unit is a two-channel—allowing maximum flexibility in stereo or mono use. The two inputs may be paralleled and each channel used separately. Channel separation is 60 dB. Weight is 105 lbs., and dimensions in its case are 40-inches high, 17-inches wide, and 19-inches deep.
Mfr: AKG
Price: $2500
Circle 58 on Reader Service Card

MINI CONSOLE

• The M675 broadcast production master when used in combination with a M67 microphone mixer provides a small size professional quality broadcast console with cuing. There are four line inputs (two switchable to RIAA). Each input is controlled individually with each pot having a cue position. An internal speaker is built into the front panel as is a front-panel headphone jack. Using with an M67, the M675 provides (built-in battery power to both units as well as facility to feed the M67 to the M675 monitoring circuits for program monitor. A rear-panel phone jack permits the user to monitor only program material.
Mfr: Shure Bros., Inc.
Price: $250
Circle 59 on Reader Service Card.

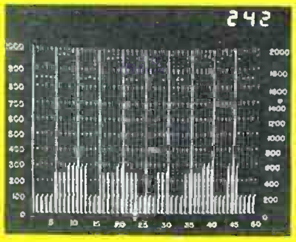
HIGH FREQUENCY HORN

• The RH60 is a glass-fiber high-frequency horn. The design uses a radial horn that has been selected and used in such a way as to provide a uniform dispersion pattern over its range in a horizontal plane. The glass-fiber material eliminates resonance and its epoxy finish will withstand considerable abuse. The horn is moulded in one piece. Its throat is 2 inches in diameter and its flange is drilled for the large JBL drivers. Adapters are available that maintain the flare rate but reduce the throat size to 1⅜ inches for Altec drivers or 1 inch for the smaller JBL's. The horn itself weighs 11 lbs. and is drilled and bracketed for easy mounting. The flare cutoff frequency is 250 Hz.
Mfr: Community Light and Sound
Circle 61 on Reader Service Card.

HIGH POWER AMPLIFIER

• Recent advances in high-voltage silicon power transistor technology is claimed as part of the reason for the availability of this new two-channel amplifier. Continuous power output per channel into 8-ohm loads is stated at 350 watts at any frequency from zero to 20 kHz. Power at clipping is further stated to be typically 450 watts continuous per channel into 8 ohms and 730 watts into 4 ohms, with power derated to 270 watts continuous per channel into 16 ohms. Frequency response is zero Hz to 0.25 MHz at 1 volt, direct coupled inputs. There are also "normal" inputs which cut off bass response at 10 Hz. Harmonic or i.m. distortions are less than 0.25 per cent, typically 0.01 per cent. Sensitivity is 1.14 V for 350 watts at 8 ohms; with hum and noise 100 dB down. Input impedance direct coupled is 10 k ohms; 100 k ohms normal input.
Mfr: Phase Linear, Inc.
Price: $749
Circle 71 on Reader Service Card.

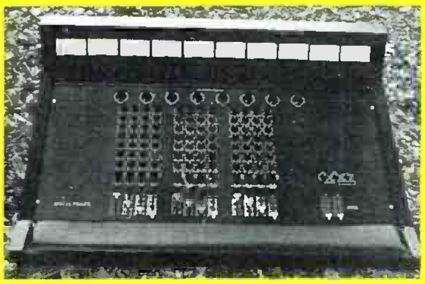
BARGRAPH DISPLAY CRT



• A new bargraph display, the MS20D-M receives digital numerical data in bit-parallel or bit-serial form, stores it in memory and displays it on a 20-inch crt. The display is an analog bargraph format. Up to eighty channels may be displayed against an electronically generated no-parallax grid. The instrument also includes provision for a decimal display of the value of the data on any channel. High and low alarm capability can be incorporated. Application as a volume-indicating multi-channel device is evident.
Mfr: Metra Instruments, Inc.
Circle 64 on Reader Service Card.

EIGHT-TRACK MIXER

• The first product of a new company, this portable eight-track mixing console is designed for studio or live performance. There are 12 input channels with 11 output busses, providing simultaneous 8 track, stereo, and mono mixdown. Each channel contains slide attenuation, e.g., stereo and echo panpots, for stereo bus selection, built-in spring stereo echo, mute switch, mic gain feedback trim, echo send, monitor level, earphone level, line/mic switch, and 8-position rotary bus selection. Eight bus output masters control the gain of each output bus. Slide attenuators are used for stereo masters. Low-noise opamp amplification, low-impedance active combining-networks, all plug-in circuit-card electronics, and a transistor regulated power supply are used. Inputs and outputs are three-pin cannon-type.
Mfr: Multi-Track Inc.
Price: $7255
Circle 65 on Reader Service Card.



NEW LITERATURE

• The De Wolfe Music Library is making available its new 500-page classified music catalog to film, slidefilm and videotape producter who want to create their own music scores for all types of a-v productions. The library itself is available on lp discs and ¼-inch tape. Of special interest is ten hours of recordings with an emphasis on the latest modern sounds of today, including electronic music. *Circle 56 on Reader Service Card.*

• Sound Reinforcement Case Histories is the title of a 12-page brochure that has been issued by Shure Bros., Inc. A total of nineteen case histories of institutional sound system applications, both permanent and portable, are detailed. Included are two churches, five educational installations, nine restaurants and showrooms, a country club, a major hospital, and a luxury shipboard installation. *Circle 51 on Reader Service Card.*

• Another Shure catalog describes the company's full line of microphone and circuitry products for the professional audio field. Illustrations, technical specifications, discussion of application and selection are included. *Circle 54 on Reader Service Card.*

• Three brochures available from Beyer describes their mics, mic stands, and headphones. One of the brochures is an article describing the falibilities of test procedures as used to evaluate the response of headphones. The others detail the specifications and prices of Beyer equipment. *Circle 53 on Reader Service Card.*

• Waters Manufacturing has issued a new four-page brochure describing their complete line of linear and rotary motion audio attenuators. Both linear and audio taper characteristic curves are offered. Specifications, outline drawings, and operating characteristic curves are included. *Circle 52 on Reader Service Card.*

• A new series of tandem (two-ganged slide switches are detailed in a new product bulletin being offered by Switchcraft. The new switches feature double-wipe slide action, for a wide variety of applications. In addition to sliders of multi-pole and throw design, several rocker-type switches are also described. *Circle 50 on Reader Service Card.*

• The 1972 Heathkit catalog now contains more than 50 do-it-yourself electronic projects ranging from a radar range, through color t.v. sets, to hi-fi equipment, and a range of test equipment of sophistication that is of value to the audio pro. For your copy, *Circle 55 on Reader Service Card.*

Circle 25 on Reader Service Card →

FAIRCHILD
SOUND EQUIPMENT CORP.
ENGINEERING DATA

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

MODULAR RECORDING CONSOLES AND INTEGRATED CONTROL MODULES

MODULAR RECORDING CONSOLES AND INTEGRATED CONTROL MODULES

INPUT MODULE, FICM
OUTPUT MODULE, FICOM
MONITOR MODULE, FICMM

With the FAIRCHILD original Integrated Control Module system you can have complete recording consoles with only three different plug-in modules, consoles with custom features that you select at the price of a "Standard" wired-in unit. Each compact "packaged" input and output module is a complete operating channel. The modules plug in (and out) as simply as a household plug, giving the units tremendous economic and service advantages.

With these unique plug-in units, a studio can be started with a few channels and gradually expanded, or channels can be moved from one studio to another as programming requires. The miniaturized solid state units provide economy and efficiency because of the close proximity of components. Down time for testing or repair is eliminated by the plug-in feature.

An unusual feature available only from FAIRCHILD is the option of eliminating functions not needed for a specific channel or application at a reduction in cost, each module being adaptable for your exact use and you pay only for the functions you need.

Utilizing the latest in solid state circuitry and the highest quality components, the FICM modules are packaged on a Formica covered strip 18½" long x 2" wide, with all input and output connections on "blue ribbon" connectors. All markings are engraved and filled for lifetime use without wear. Standard color is walnut grain, but almost any color can be supplied to match your studio decor. The modules are available individually, in "kit" form with console shell, or as a complete factory wired console.

INPUT MODULE — FICM

The FICM input module includes all necessary amplification and controls to process either mic or line level signals to mixing buss, recording machine or line for further distribution. Equalization, pre- and post echo feeds, metering and compression are included.

FEATURES

- INPUT LEVEL SELECTOR SWITCHES & PADS — either high level or mic level, both feeding into input transformer through switchable 20 db pad.
- INPUT PREAMP — with 50 db gain and +18 dbm output capability. Distortion less than 0.2% at any frequency or level below overload point.
- INPUT FADER — slide wire type provided with or without cue as desired. 125 db max. attenuation, 40 db working range. (LUMITEN light controlled attenuator available as an option.)
- ECHO FEED & ECHO FEED CONTROL, available on pre- or post-attenuator basis through pre/post selector on front panel.

- COMPRESSOR — light-activated, distortion-free, providing 40 db dynamic range, 2:1 compression ratio. 6 milliseconds attack time maximum and variable release time. Variable compression. Overload protection above normal operating levels.
- FULL SPECTRUM PROGRAM EQUALIZER — switch controlled in 2 db increments, high and low frequency response. Feedback controlled with less than 0.2% distortion. No loss device.
- OUTPUT AMPLIFIER — with 40 db fixed gain, +18 dbm output. Less than 0.2% THD.
- METERING — reference VU meter provided for monitoring input level, output level and compression. Resembles standard VU meter characteristics.
- CUE output can be provided on optional basis, allowing for audition of material processed through the module input to the fader without necessity of feeding mixing buss.
- *SPECIFICATIONS*

Maximum Gain	— 74 db (preamp gain 40 db, attenuator open, no compression)
Maximum Output	— +18 dbm into 600 ohm load, resistive or inductive
Load Impedance	— 150 ohms or higher
Output Source Impedance	— 3 ohms (without transformer)
Input Impedances	— mic input — 200 ohms line input — 600 ohms Lower source impedances can be fed directly into these inputs
Frequency Response	— ± 1 db 20 to 20kHz
THD	— 0.2% at 4 dbm. 0.5% max at 17½ dbm
IM Distortion	— 0.2% maximum
Compressor	— Dynamic range 40 db. 2:1 ratio. Attack time 6 msec (3 msec typical). Variable release time 0.3 to 7 seconds
Equalizer	— Low frequency: boost 10 db @ 50 Hz in 2 db steps droop 10 db @ 100 Hz in 2 db steps — High frequencies: 10 db boost at 2, 3, 4, 5, 7 and 10kHz in 2 db steps. 10 db droop max at 10kHz in 2 db steps.
S/N Wideband	— 70 db or better with input level −50 dbm.
Power Requirement	— 24V DC with ripple 0.5 mv p-p at 110 ma. FAIRCHILD 667II power supply recommended. Handles up to 20 modules.

OUTPUT MODULE — THE FICOM

The FICOM output module, companion to the FICM input module, contains all the components required for a high quality broadcast or recording channel. It provides necessary amplification and controls to process 0 db, — 20 db or — 40 db buss levels (so that, depending on the amount of loss encountered in the mixing networks applied, input to the module can be selected for proper signal processing), and +4 echo return level to a +4 dbm output. The module includes equalization and compression, pre- and post echo return, metering and submaster attenuator, with provision for adding an external master attenuator if desired. Output of the FICOM can be fed into unbalanced line without transformer, or a transformer can be connected internally, isolating the module from external load. Echo return input and slating input are also provided.

FEATURES

- INPUT AMPLIFIER — amplification adjustable 25 to 50 db. Output capability +18 dbm with distortion less than 0.2%.
- ATTENUATOR — slide wire type with 40 db expanded working range. Nominal attenuation 125 db. (LUMITEN light controlled attenuator available as an option.)
- ECHO RETURN — available on a pre- or post-attenuator basis, selected by pre/post switch on front panel. Echo return control selects desired amount of signal from echo or reverberation device. Accepts source impedances up to 1,000 ohms.
- COMPRESSOR — same as input module.
- FULL SPECTRUM PROGRAM EQUALIZER — same as input module.
- OUTPUT AMPLIFIER — same as input module.
- VU METER — small VU meter to monitor input level, amount of compression and output level. Resembles standard VU meter characteristics.
- *SPECIFICATIONS*

Output Source Impedance	— without transformer — 3 ohms with transformer — 50 ohms
Input Impedances	— 0 db input — 100K ohms — 20 db input — 10K ohms — 40 db input — 1000 ohms — Echo return — 750 — 1K ohms
Frequency Response	— ± 1 db 20 to 20kHz
THD	— 0.2% at +4 dbm
IM Distortion	— 0.2% maximum
Compressor	— same as input module
Equalizer	— same as input module
Wideband Noise	— 70 db or better with an input signal of — 50 dbm and +4 dbm output
Power Requirement	— 24V DC ripple 0.5mv p-p at 110 ma. FAIRCHILD 667II power supply recommended. Handles up to 20 modules.

MONITOR MODULE — THE FICMM

The standard monitor module is packaged on an 18½" long by 4" wide strip and contains provisions for monitoring ten inputs, switchable between console or tape return.

A 10 x 10 crossbar selector switch is included which enables monitoring of individual channels or any combination of channels. Gain controls for each monitoring channel are included.

Facilities for controlling slating and talkback microphones are included. Microphone preamplifier, level control and key switch for selecting either function are contained. Relay muting prevents feedback during studio recording and playback.

The FICMM monitor module requires 24V DC at 300 ma.

CONSOLE SHELLS

Shell to accommodate the 18½" long by 2" wide integrated control modules are available in unwired form, and come complete with VU meter tier, all necessary XLR connectors in the rear to accommodate desired number of channels. Channel plate for mounting mating "blue ribbon" socket assembly is included. Shells are constructed of aluminum covered in Formica to match or complement module color. Standard FICM shell is 20¾" wide by 23½" deep and contains 20" of mounting space (accommodating 10 FICM modules and/or spacers across). Other sizes available on request. VU meters are not included.

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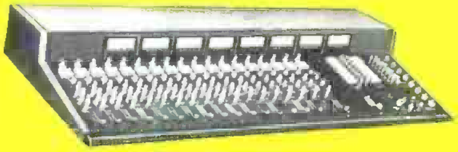
AUDIO SPECTRUM ANALYZER

● Model 8064A is a real-time audio spectrum analyzer — an instrument which measures and displays the frequency spectrum of sounds or vibrations as they occur. The unit can operate as a stand-alone analyzer or as part of an automatic or semi-automatic data acquisition, data processing, or data recording system. Applications particular to the audio profession include the ability to provide total analysis of room acoustic measurements on the spot. In short, its potential for value is so great, that we strongly suggest that you send for the data sheet that is available.
Mfr: Hewlett-Packard
Circle 72 on Reader Service Card.

18 IN, 8 OUT CONSOLE



● Reduced price without sacrificed versatility and quality is claimed for this new console with 16-track monitoring system. Consoles can be specified as to number and type of inputs, outputs, equalization, pan pots, mute and solo switches. At maximum, 18 input positions with two inputs each, mic or line, are accommodated. Minimum effort is required to change from tracking to overdubbing due to exclusive independent 16 x 4 monitoring system. Monitoring of any bus or track is possible with assignment to any or all speakers.
Mfr: Quantum Audio
Price: $5999 to $15,999
Circle 69 on Reader Service Card.

GAIN CONTROLLER

● The GC 101 wideband gain controller is a two-quadrant trans-conductance multiplier, divider, squarer, square rooter, and gain controller. It has the property of being able to control gain over a 60 dB range with 0.1 dB accuracy (1 per cent) extremely low distortion, low noise, and wide frequency response. Specifications include: response from d.c. to 12 MHz at unity gain, d.c. to 70 kHz at −60 dB gain; noise 130 mV r.m.s. in a 20 kHz bandwidth at unity gain and 6 μV r.m.s. at −60 dB gain. The unit delivers a voltage of ±10 at 10 mA. Harmonic distortion is typically 0.04 at 2 volts and does not increase at low gain.
Mfr: Burwen Labs.
Price: $250
Circle 73 on Reader Service Card.

← *Circle 25 on Reader Service Card*

DOLBY-B RECORDER

● The most recent versions of the A77 series recorders are now available with Dolby-B playback and recording circuitry built in. A total of four circuits are used, providing separate play and record facilities. The Dolby circuit is switchable in or out; there are built-in calibration controls and a reference oscillator. A built-in multiplex filter protects the Dolby circuits from interferences and inaccuracies when recording off stereo f.m.
Mfr: ReVox Corp.
Price: $200 additional to an A77.
Circle 66 on Reader Service Card.

TURNTABLE SYSTEM

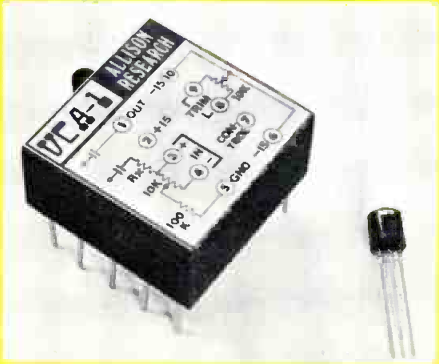
● An electronically-controlled d.c. motor is used on the model PS 600 record playing system. A relatively small motor with low dynamic mass is used to achieve low levels of motor vibration. An oil-hydraulic suspension system supports the turntable and arm mounts to reduce susceptibility to external shock. The arm is balanced in all three planes and supported by ball bearings. Specifications according to DIN standards have rumble better than −65 dB and wow and flutter 0.07 per cent or better. The unit may be used as a manual single play, automatic single play, or multiple-record changer.
Mfr: Braun
Price: $269
Circle 70 on Reader Service Card.

AUTOMATION CONSOLE

• This control center for broadcast automation has been introduced in several variations, each featuring sloping front consoles with a convenient writing surface. The control center programs music, random selects cartridge commercials, inserts station identifications, and joins networks—all automatically. It is stated to be compatible with all broadcast quality tape equipment.

Mfr: Sparta Electronic Corp.
Price: $6000-$8500 (dependent on options)
Circle 57 on Reader Service Card.

VOLTAGE GAIN CONTROL



• This potted module is designed as a voltage controlled gain device. Designated the VCA-1, it is intended for use by equipment manufacturers in forthcoming automated control boards. The manufacturer states that it solves the performance problems of conventional multiplier circuits such as poor s/n, rising distortion as gain is reduced, control-signal rejection, and low gain-tracking accuracy.

Mfr: Allison Research
Price: $50 per unit (OEM prices available)
Circle 63 on Reader Service Card.

VARIABLE FILTER

• A filter that is variable both at low and high frequencies has been introduced. The frequency points have been selected to provide an extended range of control. The low frequency 3 dB points are: 40, 55, 70, 85, 100, 150, 200, 250, 350, and 500 Hz. High frequency 3 dB points are: 15, 12.5, 10, 8.5, 7, 6, 5, 4, 3, and 2 kHz. Impedance is 600 ohms in and out; slope is 18 dB/octave, insertion loss is 0 dB in the pass band, and there is an in/out level key.

Mfr: Quad-Eight Electronics
Price: $325
Circle 68 on Reader Service Card.

CROLYN CASSETTES

• Audio cassettes using chromium dioxide coated tape are now available. On properly equalized and biased recorders, these cassettes can provide a significant increase in signal/noise and higher output at high frequencies when compared against standard ferric oxide formulations. The cassettes have been made available in C-60 and C-90 formats.

Mfr: Memorex
Circle 67 on Reader Service Card.

Martin Dickstein

SOUND WITH IMAGES

SMPTE Technical Conference

• This column will conclude the discussion on the October Conference of the Society of Motion Picture and Television Engineers. Last month, I covered the "Video Cassettes—Boom or Bust" discussion of Mr. Gordon Thompson of Bell-Northern Research Labs. of Ottawa, Canada. In addition, there was coverage of Mr. J. Bernhart of the French National Television Office of Paris who discussed standardization attempts in the low-band video recording field. We go on now to other speakers of note with the reminder that my random selection is not intended to show favoritism nor is the editing of the remarks for editorializing or personal comment or review.

In another technical session, Mr. Kazuo Iwama of Sony Corp. of America explained "The Design Concept of The Sony Color Video-cassette Total System."

Discussing the necessity of a new format, Mr. Iwama said "The half-inch format is the most familiar in the helical scan v.t.r. field. It could be very convenient for the user if the same format is maintained in the Videocassette and the player is furnished so that it can play on any standard open reel, as well as the cassette. But we have decided to develop a different format for the following reasons:

"1. Utilizing the half-inch format prevents us from designing the Video-cassette in a reasonable physical size while maintaining a reasonable playing time. 2. Designing a mechanism compatible for both cassette and open reel is complex and unpractical. 3. High density tape, recently developed, can be introduced in a new format."

The details of the Sony system are given: "Our research and development experiments led us to decide on the tape width of ¾ inch (19 millimeter). This tape width enables us to choose a tape speed of 95.3 millimeters (approximately 3¾ inches) per second, which is quite reasonable for audio. The width of video track is 85 microns (approximately 3.3 mils) and the pitch is 137 microns (approximately 5.4 mils). Both are considerably smaller, but not too small to reproduce a signal of acceptable quality for recent technology. Scanning speed is 10.27 meters/sec. (approximately 404 inches/sec.), which allows us to maintain a bandwidth of 3 MHz for luminance, and 0.5 MHz for the chrominance signal. The tape is 27 microns (1.1 mils) thick, including coating, and stored in a cassette of 140 × 221 × 32 millimeter (5½ × 8¾ × 1¼ inches) for full one hour operation. We believe that the cassette is extremely reasonable in size."

In the same technical session, Mr. Eric A. Yavitz of the Eastman Kodak Co., Rochester, concluded his talk on "Super-8 Film—A Universal Input to Video Cassettes and Television Systems" this way: "Video cassette symposia give us a chance to look into the crystal ball and foretell the future. That is always an enjoyable exercise, although there is the danger of becoming so enthralled with the future that one forgets the realities of the here-and-now, or the road that must be traveled to get from here to the future. I should like to conclude by returning to the here-and-now, and submit only one thesis for your consideration. Super 8 color film, the same film that is today increasingly widely used in the home, in industry and in the school — worldwide — is from every aspect as viable in the future world of video cassettes as it is in today's conventional display world, and perhaps even more so since it is the only system that can offer the user the flexibility of bridging the gap between these two worlds."

In a talk on "The Teldec Video Disc System," Mr. Walter Bruch of Teldec, Berlin, spoke of the Teldec equipment and the modifications introduced to the original Tripal process for color reproduction, and concluded his paper with: "The picture quality obtained with this version of Tripal is remarkable. From a normal viewing distance laymen will not observe a difference compared to a normal PAL transmission. (*Ed. note: PAL is the color t.v. transmission system used in Western Europe.*) This goes for 99 per cent of all scenes and with reproduction on a large picture tube and is due to the chosen method of composing the mixed highs. The color fidelity of pictures coming from the video disc matches with that coming from a professional v.t.r. Only the definition is given by the video bandwidth of the signal coming from the disc, but this bandwidth has no influence whatsoever on the chromaticity of the colors. The small amount of color noise is amazing but due to the color being recorded and transmitted in the best part of the video transmission range.

"Thus Tripal M or Tripal D is a method which might be applied not only for the videodisc. For instance it is excellent for video-telephone connections, or especially in NTSC countries, for the recording of television signals including color on narrow-band v.t.r.'s.

"A main advantage of the disc in comparison to helical scan magnetic video recorders is its capability to deliver an uninterrupted signal. The sync pulses for controlling a time stabilizing circuit like Ampter, can always be operational. This gives the possibility of feeding, in a later stage of development, the picture into a studio mixer." ■

PAUL C. BUFF

A Combination Limiter

The author describes both the rationale and practical considerations that went into the design of a dual limiter product that his company manufactures.

THE NEED FOR LIMITING in profesional audio work has pretty much been established and is generally understood within the industry. This article will go a little beyond the basics to more closely examine the effects of limiting on complex waveforms and on apparent levels. The main topic of discussion will be the design and use of the new Allison Research limited, Gain Brain.

I feel it necessary however, to first furnish my list of definitions of some of the terms involved:

1. Instantaneous Peak Limiter: Commonly referred to as a "peak limiter" and defined as being a limiting device which is activated by the absolute peak value (positive or negative) of the input waveform, having a compression ratio greater than 10 to 1 and an attack time of less than 100 microseconds. Release time should be long enough to prevent excessive waveform distortion. If this is not the case, I would term the device a "clipper" rather than a limiter.

2. Compressor: A peak-activated device whose compression ratio is less than 10 to 1.

3. Averaging Compressor: Generally a misnomer applied to a peak-activated compressor which has a relatively long attack time (on the order of one to ten milliseconds). While it does allow short duration transients to overshoot, it is not capable of reading the average value of a sustained waveform, hence the misnomer.

4. True r-m-s Limiter: A device whose gain reduction threshold is determined by the output of a true r-m-s detector circuit.

5. Quasi r-m-s Limiter: A device whose gain reduction threshold is determined by the output of a wave form average value detector circuit (whose output is approximately equal to the r-m-s value of the input waveform).

Paul C. Buff is president of Allison Research, Inc. of Hollywood, California

PEAK vs. APPARENT LEVEL

First, let's take a look at why limiters are used at all. I think most engineers would agree that they are primarily employed for one or more of three reasons, these being:

1. To clamp the peak excursions of a program signal to a defined electrical level so as to not cause overdriving of succeeding stages in the audio chain.

2. To enable one to record or transmit a high and constant apparent level from program sources of varying level and content.

3. To enable one to modify the sound of a program by restricting its dynamic range. (*i.e.* a "limited sound").

Let's go on to take a closer look at what's required of a limiter in order to best satisfy these objectives:

The first one is easy. We need a peak limiter with as fast an attack time as possible, as high a compression ratio as possible and release time variable to suit the content of the program.

The second situation, however, is a little more difficult. In looking at FIGURE 4, it can be seen that the electrical peak value of a waveform has no direct proportionality to its apparent level (as heard by the human ear and as read by the vu meter). If, indeed, the human ear measured the loudness of a sound by its waveform peak value, there would be no problem. We could use a peak limiter all the way around and expect to hear a constant volume regardless of the shape of the waveform. That not being the case, let's see what really happens.

If you limit, say a flute and a saxophone, to the same peak level, both your ear and your vu meter will tell you that the flute is around 6 dB louder than the saxophone. This happens simply because the saxophone has a higher peak-to-r-m-s ratio than does the flute. If you force the peaks to be equal via peak limiting (fast or slow), you cannot expect the r-m-s energy (apparent level) to be the same. It's simple mathematics. The same peak r-m-s ratio situation holds true when dealing with any combination of complex music and speech waveforms. Obviously, a peak limiter is not the answer if one is to achieve a precise uni-

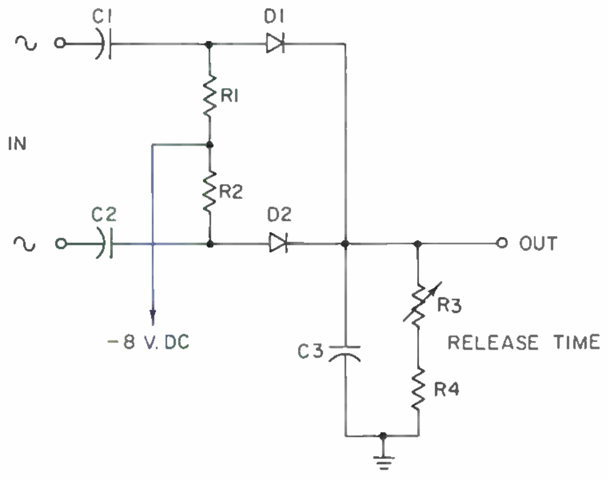


Figure 1. An instantaneous peak detector. When the input signal exceeds the reverse bias applied to D1 and D2, they will conduct—charging C3 to a voltage equal to the peak value of the input signal minus 8 volts. Speed is limited by the impedance of the drive circuit and by diode speed.

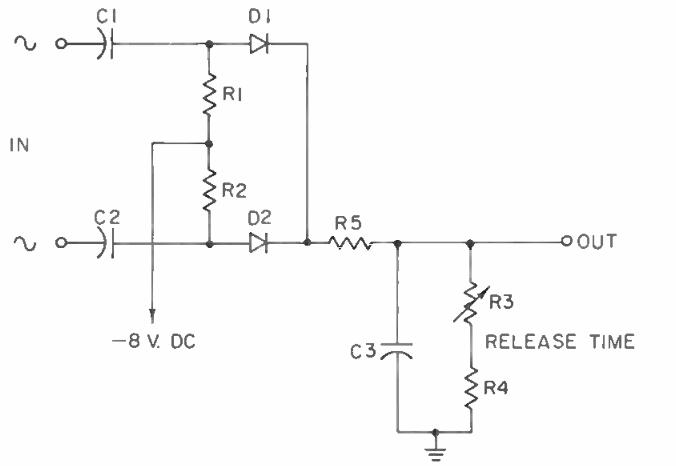


Figure 2. A slow attack peak detector. This circuit functions as the one in Figure 1, except that the speed is limited by series resistor R5. Charge on C3 will still charge to a voltage equal to the peak value of the input signal minus 8 volts.

formity of apparent level and neither is, what is sometimes referred to as, an "averaging compressor" (as described in my list of definitions.)

What is required is either a true r-m-s or quasi r-m-s limiter. In such devices, the waveform is measured in terms of its effective power (or apparent level) before reaching the threshold determining network. Only by this, or similar methods may an accurate control of apparent level be realized.

It is, however, generally necessary that such a device be followed by a peak limiter since the very nature of an r-m-s limiter is to pass peaks without regard to their intensity. The threshold of the peak limiter must be adjusted so that, while it does control the peaks, it doesn't take control to such a degree that it overrides the effectiveness of the r-m-s limiter. Generally speaking, a peak limiter threshold around 6 to 8 dB above the r-m-s threshold (with respect to a sine-wave input) will prove effective. However, certain types of program material will be found to have an optimum separation of thresholds above or below this figure.

As for the third use of limiters (getting a "sound") the type of limiting required is of course dependent on the sound which one desires to achieve. In general, the *r-m-s plus peak limiting* approach described above will enable the engineer/producer to run the gamut of limited sounds through manipulation of the thresholds and release times.

THE GAIN BRAIN

In the Gain Brain limiter, a combination of both instantaneous peak limiting and quasi r-m-s limiting is employed within the same device. The behavior of the device is similar to that which would be obtained by passing the signal first through a quasi r-m-s limiter and then through an instantaneous peak limiter. Referring to FIGURE 5, when the *function* control is rotated fully to peak position, the circuitry is such that the threshold point of both the quasi r-m-s detector and the instantaneous detector fall at the same point (with regard to a sine-wave input). Since the reaction time of the r-m-s detector is much slower than that of the peak detector, the device in this position will function purely as an instantaneous peak limiter.

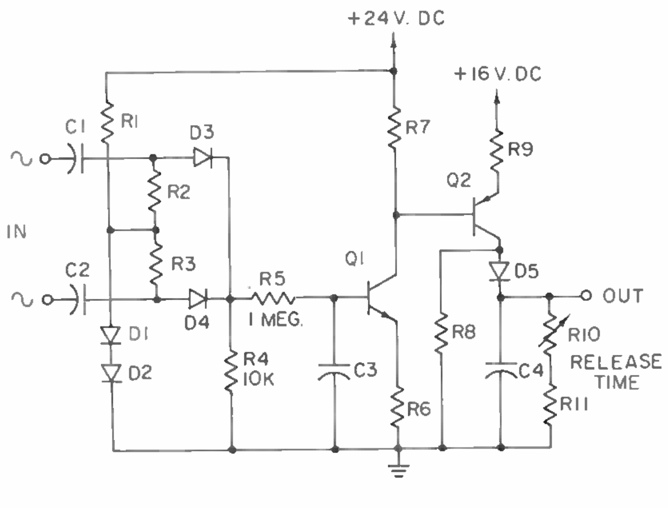
Now, if the *function* control is rotated toward the *r-m-s* position, the relative threshold of the peak detector will move up while the r-m-s detector threshold will move down. When the *function* control is rotated fully toward the r-m-s position, the circuitry is such that a 12 dB separation of thresholds will result. By adjustment of the *function* control, in this manner, the Gain Brain can be made to behave as a pure peak limiter, a quasi r-m-s limiter, or as a quasi r-m-s limiter followed by a peak limiter.

Let us examine the operation of the device when the *function* control is in mid position (6 dB separation of thresholds). If the input signal is a sustained sine wave, only the quasi r-m-s detector will react, since its threshold is 6 dB lower than that of the peak detector.

Now, assume that a sharp transient spike appears at the input; only the peak detector will react for two reasons: 1. The rise time of the spike is considerably faster than the attack time of the quasi r-m-s detector, so it will overshoot this threshold. 2. The peak-to--r-m-s ratio of a sine wave, exceeds 6 dB so the quasi r-m-s energy of the waveform is below the threshold point of the r-m-s detector.

Now, let's assume that the input is a sustained complex waveform, having a relative peak-to-r-m-s ratio of 6 dB (a typical music wave-form). Both detectors will react,

Figure 3. A simplified quasi r-m-s detector circuit. D3, D4, and Q1 are slightly forward biased by R1, D1, and D2. Voltage across C3 is equal to the average value of the input waveform, with an integration time of $t = R5 \times C3$. Q1 serves to amplify the voltage across C3. Q2 is reverse biased by 8 volts and serves as a unity gain phase inverter. When Q1 collector falls below + 16 volts, Q2 will conduct, causing D5 to conduct, charging C4. The resultant charge on C4 will equal the average value of the input waveform (times the gain of Q1) (minus 8 volts and diode drop voltages of Q2 and D5).



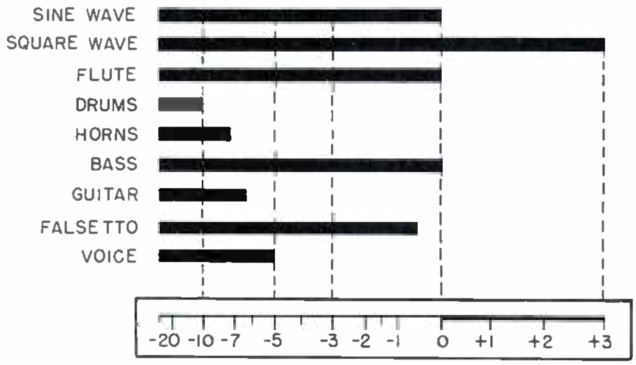


Figure 4. Typical apparent levels of various program signals whose electrical peak levels are equal.

although the peak detector will react sooner than the r-m-s detector because of the attack time differences. Should the peak-to-r-m-s ratio be greater than 6 dB, only peak limiting action will result. Should it be less than 6 dB, only quasi r-m-s limiting will occur.

RESULTING PERFORMANCE

The net result of the above, in terms of performance, is this: Gain Brain will attempt to limit the input program material according to its quasi r-m-s energy (or apparent level). However, should transient peaks or waveform peaks exceed a predetermined level (as governed by the *function* control) the device will revert to becoming an instantaneous peak limiter (for the protection of succeeding stages in the audio chain).

Front panel l.e.d. indicators are provided to inform the user which detector is being activated (both may be activated at the same time).

Generally, the most desirable results are obtained when both detectors are operating during the course of a program. Sustained simple waveforms then will cause r-m-s limiting, and complex waveforms and transients will cause peak limiting. In this manner, the apparent level of the program can be made considerably more uniform than with conventional limiters, while still maintaining an absolute control of peaks.

RELEASE TIMES

Release times for limiting caused by both detectors are coordinated and controlled by a single front panel control (*release*). In the maximum position, release is approximately 5 seconds for limiting caused by either detector. In the minimum position, release time for peak-activated limiting is 50 milliseconds, but is on the order of 200 milliseconds for limiting caused by the quasi r-m-s detector. This results in the highly desirable characteristic of allowing a fast recovery to transient limiting while maintaining a longer release time for sustained program material. Such action is desirable in that it allows the program to appear at a high apparent level without excessive low-frequency distortion and pumping effects which normally would result from fast recovery operation.

ATTACK TIMES

The attack time of the device (to instantaneous peaks) is extremely short, being less than 1 microsecond. In the laboratory, tests revealed no overshoot exceeding 1 dB within a 300 kHz bandpass, even when limiting a 50 kHz tone burst by 20 dB, with full recovery between bursts.

The circuity used to accomplish this extreme speed has still another desirable side effect. Ultrasonic transients which are less than 100 microseconds are limited with a release time on the order of 1 microsecond. This action allows such inaudible transients to be controlled without causing an audible gain reduction to occur.

The quasi r-m-s detector is activated with an attack time which is dependent on the waveform complexity but which typically falls in the 6 to 30 milisecond region.

METERING

The standard vu meter, which is commonly employed as a gain reduction meter in limiters, has some pretty serious drawbacks; the worst of these is its inherently slow speed. Simply speaking, in fast recovery limiting situations, the gain reduction has come and gone before the needle can move. This doesn't allow very accurate readings. The second drawback, is the fact that a readable vu meter just doesn't fit into a 1½-inch wide console module. (My how we've tried to make one work!)

The approach used in Gain Brain is to use a seven increment, sequentially activated, l.e.d. array to accomplish gain reduction metering. Inherent l.e.d. speed is measured in nanoseconds. Readability is at least equal to that of a 4-inch illuminated vu meter. Panel space required is approximately ¼ by 2-inches. Thanks to the sequential operation of the increments, an array of such meters does not cause the distracting light show effect or high current consumption of certain non-sequenced beam of light meters (where as many as 9 lights per meter may be on at one time). A further highly attractive feature of l.e.d's is the fact that the life expectancy of these devices is the same as any other semiconductor—virtually limitless. ■

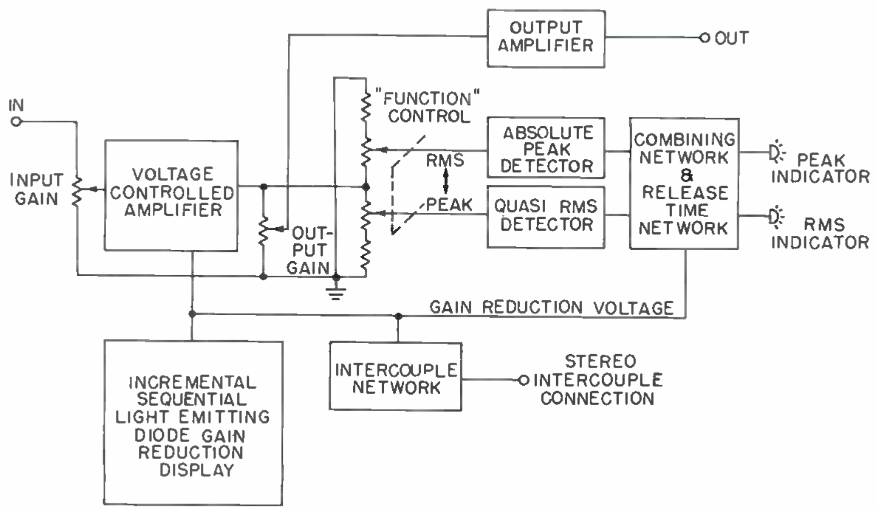


Figure 5. A simplified block diagram of the Gain Brain described in the text.

ROBERT C. EHLE

Circuits for Synthesizing Multi-Channel

In this article, the author describes experimental circuits that may be built from readily available parts. They may be used to synthesize additional channels from one- and two-channel sources.

THE CIRCUITS that are to be described include a method for synthesizing two-channel stereo from a mono source and several different techniques for producing four-channels from a two-channel source. Techniques for synthesizing four channels from a mono source will also be discussed and illustrated and some methods for using more than four channels will also be mentioned.

SYNTHESIZING TWO CHANNELS FROM MONO

Stereo synthesizers have been built for years using the band-pass technique. In this circuit, several band-pass filters are connected so that alternate bands in the frequency response are delivered to the left and right channels. Probably the simplest experiment in this area is to turn up the bass and turn down the treble on the left amp and set the reverse on the right amp. Commercially-available stereo synthesizers (Orban Parasound, Countryman, etc.) utilize several bands (typically five) with bands *one, three* and *five* passing to the left channel and bands *two* and *four* going to the right. These commercially-available synthesizers are frequently used to "enhance" mono recordings for release in synthetic stereo format.

The circuits of commercially-available stereo synthesizers typically utilize active devices such as integrated circuits as well as resistors and capacitors of precision values. There is another method, generally handier for the home constructor, which utilizes common nonprecision sizes of inductors and capacitors. This is the circuit described here. Another advantage of the circuit is that no power supply or active devices are required in normal usage.

The circuit designed by me and illustrated in the schematice diagram of FIGURE 1 uses series resonant circuits made up of paper capacitors and iron-core inductors. The circuit is inherently a low-impedance device and works well between typical transistorized preamps and amplifiers. If high-impedance circuitry is used, as would be the case with most vacuum-tube amplifiers, parallel resonant circuits in series to ground should work as well, although this has not been tried by me. Isolation of the outputs may pose a problem unless balanced inputs and outputs are used.

Writing by designers of the commercially-available stereo synthesizers has emphasized the matter of phase shift in the performance of their units. The desirable situation would seem to be that no signal should come from the two speakers with the same phase, and signals coming from the two speakers with the same amplitude should be 90 degrees out of phase. It is worth noting that this is approximately the case with the present circuit as the phase angle of the output current in each band will be zero degrees at resonance and will lead up to 90 degrees below resonance where it is capacitively controlled, and will lag up to 90 degrees above resonance where it is inductively controlled. The amount of lead or lag is directly related to the amount of attenuation so that the point where two adjacent bands cross the upper band will have a leading phase and the lower one a lagging phase—thus approximating the desired situation.

The precise phase relationships can be worked out mathematically considering the number of bands, the Q of the resonant circuits, the resulting slope of the filter bandpasses and so forth, but this would require the purchase of unusual precision-value components. The component values given are common and work well in actual listening tests. One word of caution is necessary concerning the inductors: avoid swinging chokes. Audio transformers work well and high Q inductors (such as toroids) are even better. The experimenter can try inductors he has on hand by connecting them temporarily and running a frequency-response test with a signal generator and a meter to determine the resonant frequency. In general, inductance will be found to be directly related to the size and weight of iron-core inductors due to the predominate effect of the core. Parts values for the synthesizer are generally not very critical and good results will be had by selecting components in the general range of those given—just making sure that the sizes are ordered from largest to smallest as shown.

SYNTHESIZING FOUR CHANNELS FROM TWO

Most present readers are surely familiar with the Dynaco systems of synthesizing four-channel stereo from two channels. One such system, designed and described in the literature by David Hafler, make use of two additional speakers connected to the stereo amplifier, one sum connected

Robert C. Ehle, Ph.D. is with the University of Northern Colorado's School of Music. He has written extensively on electronic music subjects.

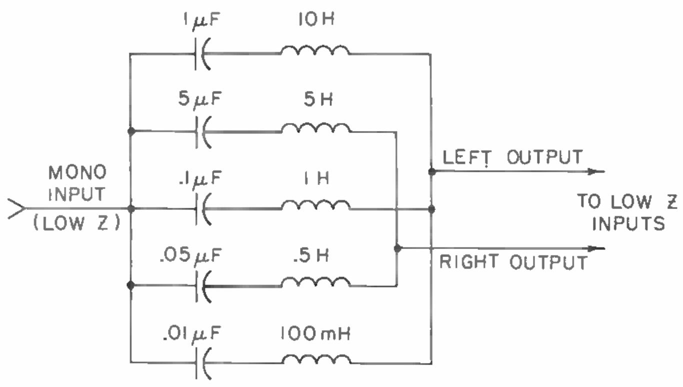


Figure 1. A stereo synthesizer designed by the author. Frequency bands for the five filters are centered on the following frequencies (top to bottom): 50 Hz, 318 Hz, 500 Hz, 3180 Hz, and 5000 Hz.

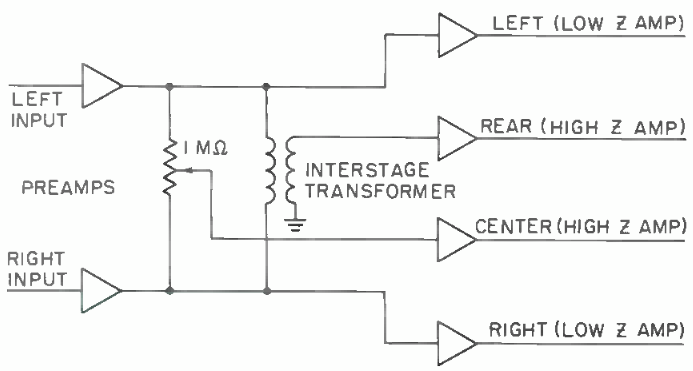
and the other difference connected. Dynaco control units are also commercially available to make this connection.

Perhaps some experimenters have four amplifiers available and would like to make the sum and difference connections at the preamp outputs. The circuit described here is intended to perform the same function as the Dynaco system but be connected at the output of a stereo preamp and before the four amplifiers and speakers. In its simplest form, as I have drawn it in FIGURE 2, two of the amplifiers, the sum and difference channels, must be high input impedance amplifiers such as vacuum-tube types. This may prove particularly convenient for others as it has for me, because many people may well have two vacuum-tube amplifiers available that they have replaced with transistor types. This circuit allows them to be put back into service.

The circuit works as follows: the sum channel is derived by connecting a 1-megohm potentiometer across the two-channel output and the sum taken from the slider. This may be balanced for equal amounts of the two channels in the sum output. The connection must be to a high-impedance input or excessive power would be developed across the resistor and none would reach the amplifier. A high-impedance resistor is required to prevent loss of separation between the two low-impedance preamp outputs. The right and left amplifiers should be low impedance input types such as is typical with transistor types.

The difference output is developed with a transformer.

Figure 2. In this four-channel 1-to-two synthesizer the interstage transformer should be 10,000 ohms primary and 600 ohms secondary. This type of audio interstage transformer is commonly used to couple a plate output to a 600-ohm line. When the 600-ohm output is connected to a high input impedance amplifier, the reflected impedance becomes quite large. This is to prevent a loss of channel separation.



Current will flow in the primary only when the right and left channels carry different voltages. This "difference" signal is transmitted to the secondary where it is sent to the difference or rear speaker. This rear-channel amplifier must be high impedance so that a high reflected impedance is seen by the two inputs. This prevents loss of separation.

Several variations on the four-from-two circuit are possible. One of these is to replace the rear speaker with two side speakers hooked in parallel. There may also be two side amplifiers if this is convenient as it makes certain other experiments possible. This is the connection of the rear channel recommended by Madsen, Cooper and others following the Damaske effect which shows that side speakers are about 26 dB more detectable than rear speakers when presenting ambiance information. This allows these speakers to function at lower levels while providing a smoother rear-speaker effect. Another advantage is that the two rear speakers and amplifiers are available for discrete four-channel operation in their accepted positions of left rear and right rear.

ANOTHER METHOD FOR SYNTHESIZING FOUR FROM TWO

A method for synthesizing four channels from two but with a slightly different effect than the previous circuit may be tried by building two of the stereo synthesizers described in the first part of this article and hooking them up so that the right and left rear channels are synthesized from the right and left front channels. The front channels contain the unmodified stereo signals. It is shown in FIGURE 3.

This type of circuit does not enhance ambiance information the way the Dynaco circuit and many other stereo synthesizers do, but it creates four unique channels, each containing a portion of the stereo signal in greatest prominence. In other words, rather than synthesizing an auditorium and putting the listener in the ideal seat, it synthesizes a very large musical stage and puts the listener in the middle with sound sources coming from all directions. Some listeners may find that they like this effect very much; it is particularly interesting with modern music as it clarifies the texture considerably.

SYNTHESIZING FOUR CHANNELS FROM MONO

Four channels may be synthesized from a mono source by using a combination of the circuits just discussed. If a mono signal is fed to the stereo synthesizer and then the outputs are used to feed right front and left front, center and rear channels may be derived by using the resistor and transformer circuit. It is important to remember, though, that this method will not simulate ambiance. It can only produce a spatial distribution of the various frequency components of the signal combined with phase shift as previously discussed. The only method shown to be useful in synthesizing ambiance information with mono sources is a time delay or a reverberation device.

A circuit has been devised by me (FIGURE 4) that makes use of two small spring reverb unit, a single driver amplifier, and a stereo synthesizer (as described previously) to synthesize four channels from a mono input. The spring reverbs are manufactured in Japan and sold in the U.S. by the Olson company of Akron, Ohio. The units have the stock number of X-75 and cost $1.39 each. The driver amp must have low-impedance outputs; the unit used by the author is a modified automobile radio. With a mono input, only a mono driver amplifier is required; two reverb units were used to keep the two rear channels from having exactly the same output. Both are driven from the same amplifier.

The effect of this circuit is peculiar and not entirely natural. The artificial reverberation is much more pro-

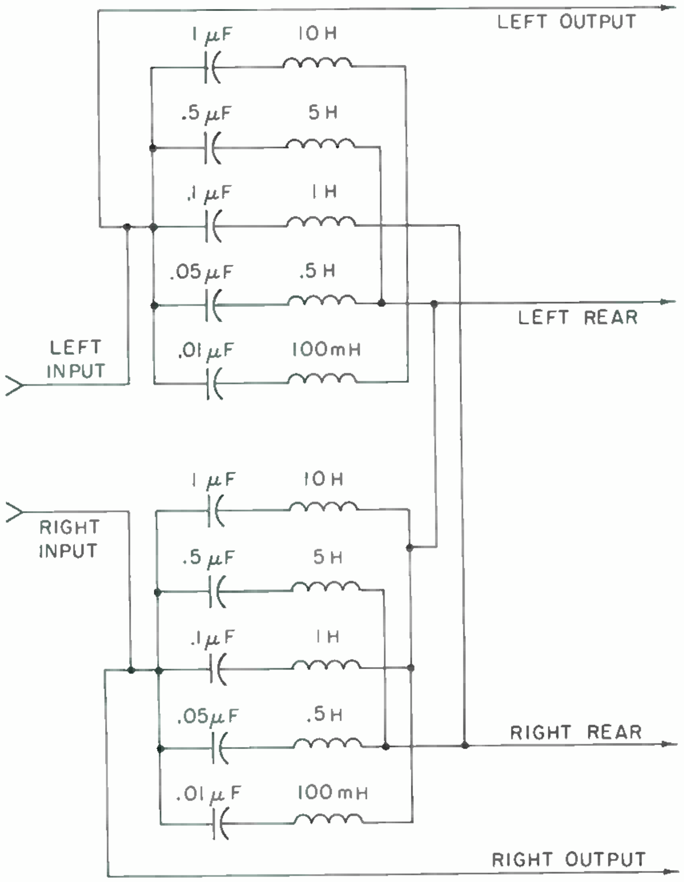


Figure 3. This circuit uses two stereo synthesizers to simulate four channels from normal two-channel stereo.

nounced on voice than on music. It is only slightly detectable on classical music where the effect is close to the desired one. Speakers take on a rain barrel effect which is very noticeable and usually unpleasant. The spring reverb system may be used to produce some novel, "far out" effects as well, but generally it will not be very desirable for extended listening due to the common drawbacks of this type of unit: strong resonances, flutter, narrow frequency response and sensitivity to shock and sudden transients. The spring reverb system may be used with stereo signals as well by deriving its input from a sum connection such as the potentiometer circuit described previously. In cases of this sort, it is usually preferable to use the sum

Figure 4. This experimental circuit can be used to synthesize four channels from a mono input.

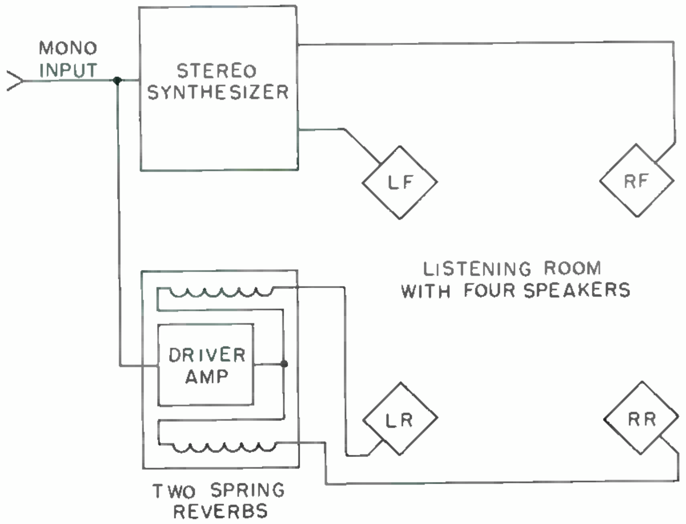


Figure 5. The author's experimental electronic music studio. The basic synthesizer, tape recorders, and ancillary equipment is evident.

and difference circuit and to try to extract natural ambiance information from the recording. Other reverb and delay systems may provide a more musical result and are worth investigating. The tube reverb system should avoid many of the problems associated with springs; the sheet metal reverb does eliminate some of them (flutter, strong resonances), and there are high-priced digital delay systems on the professional audio market that should do everything for those who can afford it. Another possibility is a high-speed tape delay with closely spaced record and play gaps. The delay should be about 10 milliseconds.

MULTI-CHANNEL SYNTHESIS FOR ELECTRONIC MUSIC

While it is generally recognized that the purpose of most multichannel setups is to simulate or to reproduce music as it should sound in the optimum concert hall, this is not at all the case in electronic music. Here the spatial dimension is just one more parameter at the disposal of the composer who may utilize it as he wishes. In general, it is likely that he will not care to simply synthesize a concert hall for his music to sound in. He will, very likely, want to utilize moving sound sources, total surround sound and larger than life spatial dimensions. Thus he is likely to use all of the techniques listed here, including those having exaggerated effect and many more not listed here. His basic problem is similar to that of the average music listener only more acute: how can he get complete control over four channels of information using only two channels for transmission? Since he requires more than simple ambiance information, the demands he would make on the system are greater than those of concert-hall simulation systems. But, since nearly all the outlets for his product use two-channel records or tape he is generally constrained to reduce his four-track master to a two-track copy for records and radio.

For all of these reasons, the ultimate experience in electronic music is not likely to be obtainable at home from radio or phonograph records but only in specially-prepared listening rooms or studies such as that at the Belgian Worlds Fair where Varese's *Poeme Electronique* was presented in sixteen channels, or at the Electric Circus in New York City and several of the world's electronic-music centers. Here the composer is free to work without excessively constricting technical limitations and the results are quite unlike music heard elsewhere. ■

RICHARD L. LERNER

A Frequency Counter Meter

Frequency counters are a useful tool to the broadcaster and recording engineer. In fact, virtually anyone who has ever to know the identity of a frequency can find value from such a tool.

THERE WAS A TIME when the measurement of the exact frequency of an audio or radio frequency signal was at best a guess. Individuals of limited means could rely only upon comparison with signal sources which were highly innacurate. Comparison by means of Lissajous figures on an oscilloscope, involving counting the ratio of lobes on the vertical and horizontal sides of the figure, were tedious, time consuming, and frequently in error. There were also horrendous gaps in the available range of usable calibration frequencies for checking the results, and interpolation to find frequencies in between those which were integral ratios of the available signals was best done with the fingers tightly crossed.

Fortunately for us, the world of electronics has come a long way since then, and the digital electronics frequency counter has been invented. Furthermore it has been highly refined, and greatly simplified. Products have been on the market for some time now in the under-a-thousand-dollar-range which make the accurate measurement of signal frequency a matter of simple connection to the signal and reading off the result. Most of the available devices are intended for measurements in the 10 Hz to 15 MHz range and operate satisfactorily with inputs of as little as ten millivolts.

Recently, the availability of a low-cost electronic frequency counter kit was announced by the Heath Company, In Benton Harbor, Michigan. They describe the unit as a compact, lightweight, rugged digital frequency counter, having five cold cathode display tubes, an overrange lamp, and two range indicator lamps. Modern digital techniques and the use of integrated circuits and f.e.t. transistors (almost exclusively) have resulted in a neat, compact instrument which wegihs only 4½ pounds and measures only 8¼-inches wide by 3⅜-inches high by 9-inches deep. Heath also makes the instrument available in a factory-wired version, and either type is adaptable to 105-125 or 210-250 volts operation. It is intended for use in ambient temperatures between 0 degrees to + 40 degrees Centigrade, but can be stored in environments ranging from — 55 degrees to + 80 degrees Centigrade.

The kit version is typical of Heath, which is to say it goes together relatively easily, and is supplied with all materials needed to do the job, including a more-than-adequate instruction manual on general electronic assembly and wiring techniques as well as the expected assembly and operation manual for the counter itself. A check of the parts supplied against the parts revealed no shortages, and no damaged parts. In fact, the parts were protected against shipping damage very well, particularly in the case of most of the integrated circuits, which were placed into individual plastic receptacles. All of the electronic parts, with the exception of the power transformer and the front panel controls and light, go onto a well laid out fiber glass printed-circuit board. It does seem at first glance that all of those parts will never fit on approximately fifty square inches of board, but fit they do, neatly and clearly marked for identification. The accompanying photographs can only give an idea of what is involved in building the kit, but with one minor exception, the kit was a pleasure to assemble. That one area lies under the Hz/kHz indicator lamps, and it is quite possible to either burn parts while wiring or to build in an accidental short circuit. While we did neither, it is one place where care is advisable. Of course, care should always be taken in soldering printed-circuit boards, particularly those which have parts spacings as close as those required by integrated circuits. Heath has, however, removed one critical soldering task from this kit, and they should be commended for it. Anyone who has had to remove a "dead bug" (defective integrated circuit) will attest to the convenience of i.c. sockets, to say nothing of the significantly lower risk to the i.c.'s themselves when they are not soldered. The sockets do require a bit of care in placement. They are made of strips of connector pins which are then cut to the appropriate lengths, placed, soldered, separated from their connecting bands, and then receive the i.c.'s Here also, care is required, since it is easy to fail to insert one of the bug's legs. The integrated-circuit parts took about 1½ hours of the total six hours necessary to build the kit; working slowly and carefully.

Before going into what the Heath counter can and cannot do, perhaps it would be well to discuss counters in general and to define a few terms. A digital counter operates on the principal that it can be told to count incoming signal frequencies for a known period of time and display the attained total in terms of signal frequency. From this it can be seen that the indicated frequency count can be no better than the accuracy of the time that the counter is gated to count. An additional inaccuracy lies in the fact that most frequency counters do not have synchronization between the gating clock and the incoming signal. Thus, it is possible to have the last pulse of the counted period either just registered or just not registered. For this reason,

Richard L. Lerner is assistant editor of **db Magazine.**

any frequency measured must be considered to have a plus or minus one-count inaccuracy. For the same reason, the first count of a signal under changing frequency conditions is usually very wrong, since there is no telling where the count started with respect to the cessation of the frequency variation. It is therefore necessary to have the signal being measured remain stationary for the period during which it is being counted. The counter cannot be expected to perform accurately in the presence of noise which might be counted as though it were signal, although in general quite a bit of noise and distortion can be tolerated compared to that which would generally make the signal difficult to use for audio or r.f. purposes.

The accuracy of the frequency indicated, assuming that the input signal is stationary, is then influenced by several factors within the counter itself. These are the inherent accuracy of the time base, the tendency for the time base to drift with time, and the effect of temperature on the time base. Of them all, aging is the least considered, and generally runs in the order of one part per million per month in inexpensive counters. Obviously, this doesn't make very much difference for most measurements, and can be left out of calculations of accuracy if the instrument is calibrated against a known standard about every six months. When a counter is calibrated and used in a laboratory, the effect of temperature can also be ignored, but should be considered in the case of the heater or air conditioner that fails just when important measurements must be made.

The inherent accuracy of the time base turns out to be the largest influencing factor in the frequency-measurement accuracy. Perhaps the best source available generally is radio station WWV, maintained by the National Bureau of Standards for just such purposes, broadcasting at several frequencies which are exact multiples of 5 MHz. Other sources of a standard frequency are available, but it should be remembered that no measurement can be more accurate than its initial calibrating standard.

Some counters can be used to do more than count frequency. If the gate can be controlled externally, the

Definition of Terms Used In Judging Frequency Counters

Accuracy: The capability of an instrument to ascertain the true value of a measurement. Frequently used instead of innaccuracy, which is the value by which an instrument reading departs from the true value and into which all causes of error are summed.

Calibration: The comparison of actual readings made by an instrument with previously established standards. The deviation of the indicated values from the standard is used to infer the true value from the indicated in making measurements. This term is frequently used interchangeably to mean the process of adjustment required to fix, reduce, or even eliminate the deviation.

Digital Reading: A measured value in terms of discrete magnitudes as opposed to an analog measurement, which can have an infinite number of continuous readings.

Distortion: An unwanted change in the waveform of an electronic signal. Used generally to describe non-linearity of a device which can cause a large source of measurement error. Modulation is a controlled, desirable form of distortion.

Gate: An electronic circuit which is designed to pass signals for only a selected fraction of a predetermined time interval.

Noise: Any unwanted electrical signal which disturbs the measurement of data.

Operating Temperatures: The temperature or range of temperatures within which an instrument can be expected to operate within its rated limits of error.

Period: The time required for one complete oscillation or a complete event cycle. The reciprocal of frequency.

Precision: A word all too often confused with accuracy, meaning the intended or design performance rather than actual performance. It is synonomous with resolution in the sense of readability, sharpness, or clarity.

Repeatability: The maximum difference in readings for a given set of identical measurements.

Resolution: The smallest change in the quantity being measured which produces a change in the instrument reading.

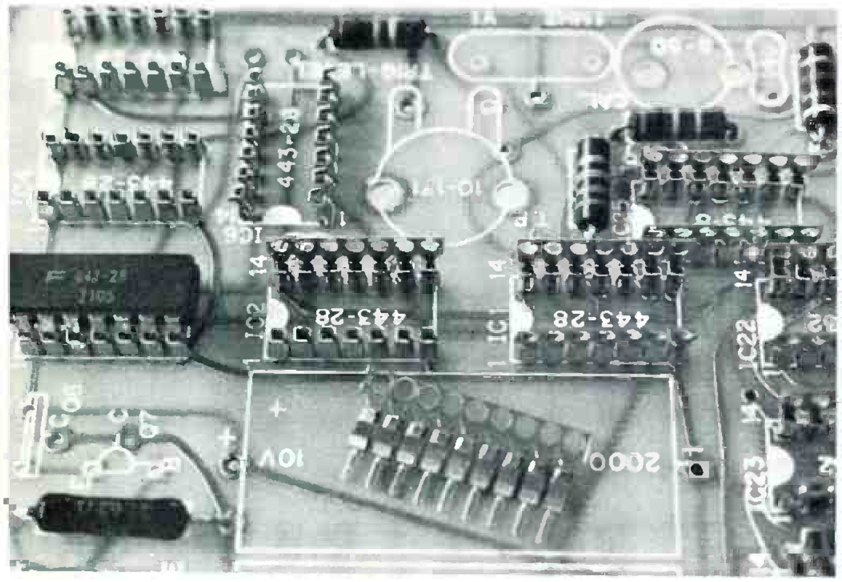
Sensitivity: The amount of signal required to produce a unit deflection of an instrument. Usually expressed in terms of the maximum sensitivity, or the least scale factor response. In a counter, the lowest input voltage which permits triggering.

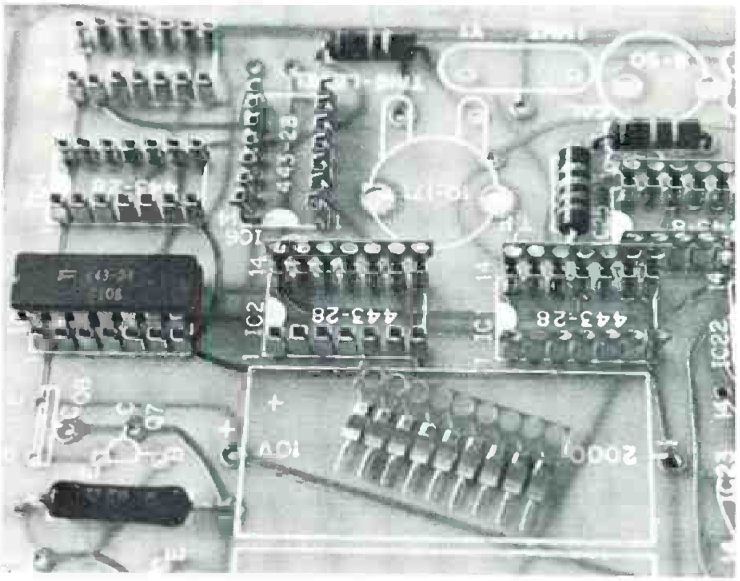
Stability: The absence of drift, or more broadly, freedom from change in output from causes other than input changes.

Triggering: The operation of the counter's gating circuits.

Time Base: A reference time signal used to establish the time relationships of signal events. The heart of a counter.

How the Kit Went Together



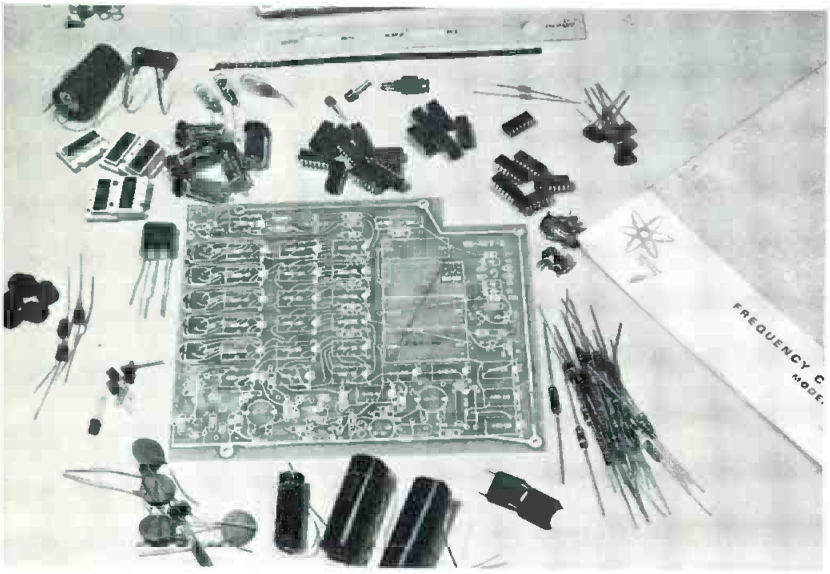


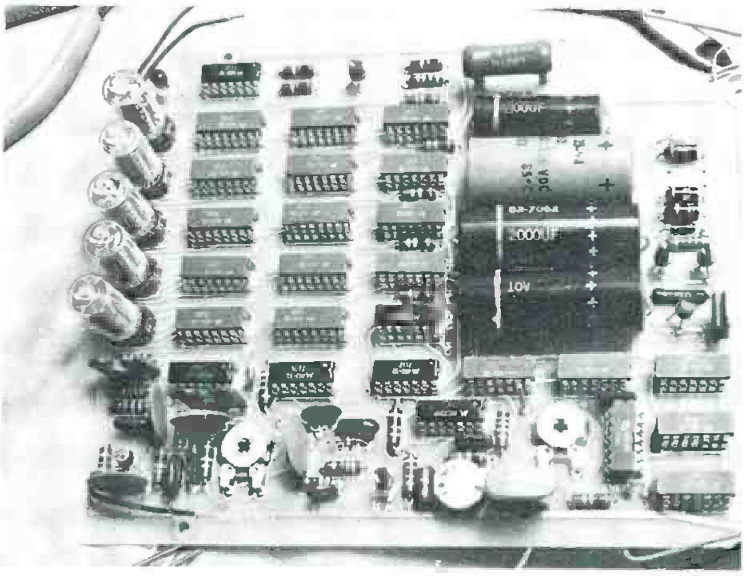
The componentry unpacked. Note the circuit board on which all the electronics will go, save only the power supply.

The circuit board and its hardware, and components. Clear instructions are presented in a logical fashion.

The board during construction showing the installation of the i.c. connectors, with one i.c. already in position.

The completed circuit board. Note the readout tubes on the left, this will be the front of the finished unit.



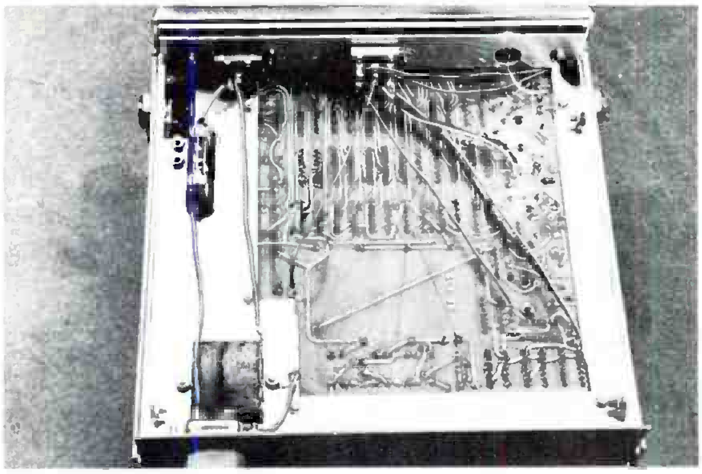


counter can rapidly count for periods which are a function of anything the user desires. For example, an open gate would permit a continuously running total. Counting the frequency of a known source while controlling the gate start and stop permits timing the period between start and stop to great precision. Some counters are available for totaling in both directions.

The Heath unit however, is not intended for any of these specialized functions. It is intended for and performs well in the range of 1 Hz to 15 MHz, with gate times of either one millisecond or one second, and will operate to count signals in that range with a sensitivity of less than 250 millivolts. It will tolerate r.m.s. voltage inputs of 200 volts up to one kHz, derating smoothly to 5.6 volts at its upper frequency limit. Trigger levels are internally set automatically, and the input impedance is one megohm shunted by less than 20 picofarads. The time-base frequency is crystal controlled at one MHz, with sufficient adjustment available for calibration. Heath states an aging rate of less than one part per million per month after the first thirty days, and 0.2 parts per million per degree Centigrade for normal operating temperatures.

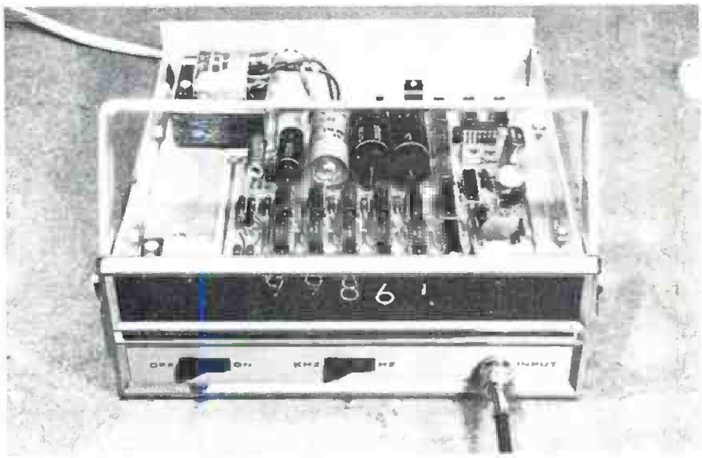
It is possible to use this counter, and most others, to cover ranges of frequency far greater than the rated coverage. There are digital devices available which divide the incoming signal by ratios of ten before presenting it to the counter. It is frequently possible and generally simple to mix the unknown high frequency with a known high frequency source, such that the counter can be used to count the difference or beat frequency. This method is obviously highly dependent upon the qualities of the beating signal source.

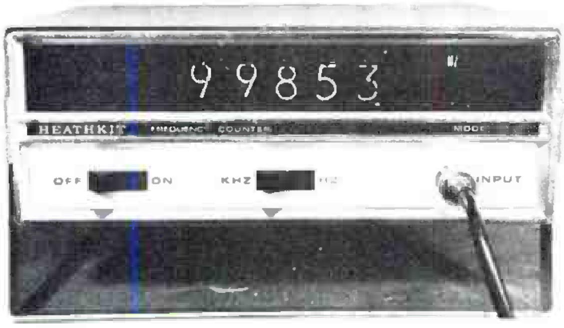
The Heathkit counter has one other interesting feature, in common with most similar units. When a frequency higher than its Hz range is being counted, it displays the residual count. Thus, all eight figures of a 15,000,000 Hz signal can be resolved, by displaying the count on both ranges. The use of these figures should include due care in considering the relationship between precision and accuracy. ■



The finished kit from the underside after the circuit board and the power supply have been brought together.

Undergoing initial setup and tests. The uncluttered finished appearance is apparent. Note the bnc connector.





Completed. At the left of the display a lit indicator tells when an over-frequency is present.

● Just as we were going to press with this article we learned of the addition of new counter models to the Heath line. The IB-101 discussed remains available as described, but an IB-1101 model is now available with a counting range of 1 Hz to over 100 MHz. This is available as a kit with similar assembly procedure to the described model. In addition to this new kit, Heath has also announced two assembled models, both available only as factory wired units. Both count from 10 Hz to over 80 MHz. The SM-104A and SM-105A models differ only in stability and time base accuracy with the SM-104A providing a 1 MHz time base that is ±0.1 Hz with a ±1 ppm/year stability; the SM-105A uses a 1 MHz crystal accurate to ±2 Hz, with an over-all time base accuracy of ±10 ppm. Both use l.e.d. readout devices.

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PEOPLE, PLACES, HAPPENINGS

WORTMAN

• **Leon A. Wortman** has been elected western regional vice president of the **Audio Engineering Society** for 1971-72. He is the manager of corporate marketing services for **Ampex Corporation,** Redwood City, California.

As one of three regional vice presidents of the AES he is responsible for coordinating society programs in the western states. He also is in charge of arrangements for the AES convention in Los Angeles in May.

• The **Holiday Inn** in Munich, Germany, will be the locale for the **Second Annual Convention** of the **Audio Engineering Society Central Europe Section,** to be held March 14-16, 1972. The occasion will again provide a forum for professionals in audio engineering and electroacoustics from Europe and abroad. Papers in both these fields will be presented. This year will also feature exhibits of the latest professional products by manufacturers in audio and the allied arts. A number of American exhibitors are expected to participate. For exhibit information, contact: **K. J. Wischgoll, E. Byer Elektrotechnische Fabrik, D-7100 Heilbronn, West Germany.**

• The **42nd Convention** of the **Audio Engineering Society** will be held at the Los Angeles Hilton Hotel, Los Angeles, California, May 2-5, 1972. Technical sessions on new developments in the field of audio engineering will be offered, concurently with exhibits of professional audio equipment. Send titles and abstracts to: **Leon A. Wortman, Ampex Corporation,** 401 Broadway, Redwood City, California 94063. For exhibit information, contact: **Jacqueline Harvey,** Exhibit Manager, Audio Engineering Society, 124 East 40th Street, New York, New York 10016.

• The president of **Electro Sound, Inc., Allen Weintraub,** has announced that through their exclusive distributor, **Audiomatic Corporation** (whose president is Milton Gelfand), a sales contract has been entered into with **Melodia Records,** U.S.S.R. Melodia had negotiated with Audiomatic Corporation for installation of complete turnkey operations.

The president of Melodia Records, **Vasily Pakhomov,** and the chief technical expert, **Dmitri Kazman,** had visited Electro Sound, New York, and **E. Lewin,** vice president of marketing for Electro Sound, escorted them on a tour through Elctro Sound's plant in Sunnyvale, California. After returning, Mr. Pakhomov indicated to Mr. Gelfand that the visit to Electro Sound's plant had convinced him that Melodia Records will supply all of their plants in the U.S.S.R. with Electro Sound high speed duplicating systems and peripheral equipment for complete plant operation. The first sales contract is in the amount of $350,000.00 and the equipment will be shipped in the beginning of this year.

In addition to the duplication equipment, Audio Matrix, Inc. (Mr. Gelfand is president of this company) is selling an eight-position Audiomatic Process record plating system to Melodia. The sale is the first such provided by an American company to the Soviet record industry.

• The election of **Adolph Wolf** as president of **Electro-Voice, Incorporated,** a subsidiary of Gulton Industries, Inc., was announced today by **Walter F. Gips, Jr.,** Gulton president and chief executive officer. **"Mr. Lawrence L. LeKashman** who had been president of Electro-Voice, has resigned to accept the position as president of **Olson Electronics, Inc."** Mr. Gips stated. "While we regret Mr. LeKashman's decision, I am confident that Mr. Wolf will maintain the leadership and continue the growth of Electro-Voice in the electro-acoustic industry." Mr. Gips continued, "He has had many years of key management experience with Electro-Voice and, previously, with Webcor and Zenith."

• The board of directors of **Ampex Corporation** have announced the resignation of **William E. Roberts** as president and chief executive officer and the election of **Arthur H. Hausman** to succeed Roberts in these posts. Roberts, president and chief executive officer since 1961, continues as chairman on the board of directors of Ampex. Hausman formerly was chief operating officer and executive vice president of the company.

• **Stephen F. Temmer,** president of **Gotham Audio Corporation,** New York and Hollywood, takes pleasure in announcing that **Robert Buescher** has joined the sales staff. His major responsibility will be the marketing of the **Stellavox** Portable Tape Recorders. Mr. Buescher was formerly the sales manager of **Nagra Magnetic Recorders, Inc.** He has been associated with the professional film recording field for 21 years as a sales manager for **RCA Films** and as the chief engineer for **Recording Studios, Inc.** in New York.

• **Mr. K. Tani,** president of **TEAC Corporation of America,** announced that the facilities of the new TEAC subsidiary, **TEAC Special Audio Systems Corporation,** are now nearing completion and will be in full operation by February 1972 with an initial complement of over 150 people. The subsidiary will be based just outside Tokyo at Kodaira City. Mr. Tani stated, "This new facility will be devoted to the manufacture of special audio products which will be used by professionals such as broadcasters and audio studios. This plant will specialize in products not being mass produced which will enable us to provide sophisticated high quality products and will incorporate the skills and technical know how that TEAC has acquired in audio and magnetic tape products. It is expected that this plant will produce such items as broadcast consoles, studio mixing consoles and many other similar products. It will also have facilities for the manufacture of pre-recorded tapes which will include a variety of special test tapes."

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And that's where our new Revox A77/ Dolby B recorder comes in.

By now, the virtues of the Dolby Noise Reduction system are too well known to require any elaboration on our part.

Suffice it to say, for all practical purposes the last major stumbling block to quality, noise-free recording has finally been eliminated.

Listening to tapes on the new Revox/Dolby B is a revelatory experience. Tape hiss is virtually non-existent. The music seems to emerge from a background of velvety silence. And at 3-3/4 i.p.s. the absence of extraneous noise is truly startling.

But no mere description of the Revox/Dolby B can adequately convey the experience awaiting you the first time you listen to a tape made on this remarkable machine.

Your nearest Revox dealer will be delighted to audition the Quietest Revox for you. Once you've heard it, you'll understand why we say...

Revox delivers what all the rest only promise.

The Revox/Dolby B

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