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THE SOUND ENGINEERING MAGAZINE

JUNE 1969 75c

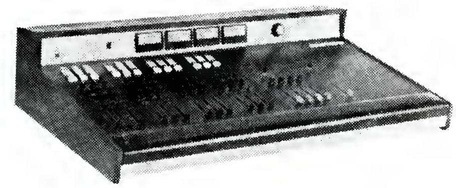
The Ultimate Noise

Implications of the Low-Noise Background

Multi-Purpose Halls Present Problems

Picture Gallery — 1969 NAB Convention

NBC nixed our audio control console...



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Coming Next Month

●A special issue devoted to the subject of television sound.

Marshall King of the Hollywood Video Center where he mixes such shows as the Steve Allen Show and the King Family Show has contributed an article on TV AUDIO MIXING that details the efforts to achieve a first-rate sound track on the video tape that leaves his studio.

Meanwhile, back East, the New York Chapter of the AES invited several TV audio mixers to a discussion of this topic. Our tape recorder was there and we have a partial transcript of the lively discussion that ensued.

Learn what happens between the lovely sound of the studio and the final product that the listener/viewer gets at home.

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

About the Cover

●A montage of booths seen at the NAB Convention held this past March in Washington, D.C. See page 33 for the picture gallery.

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THE SOUND ENGINEERING MAGAZINE

JUNE 1969 • Volume 3, Number 6

Table of Contents

FEATURE ARTICLES

MONTHLY DEPARTMENTS

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Letters

The Editor:
The engaging Edward Tatnall Canby, who has stimulated almost a whole generation of audio professionals and almost singlehandedly developed an informed audio laiety, does an injustice to the U.S. system of television. Oh, what we seen on the usual set is every bit as deplorable as he describes it, in his article in your January issue, but it's *not* because of "our present transmission standard." It's because of shortcuts in our receivers, presumably the necessary consequence of competitive prices. Otto Schade of RCA established early in television's history (and reported in a classic series of SMPTE papers) that the 525-line 4.25 mHz system can produce pictures *of approximately the same subjective quality as projected 35 mm movies.* Hardly any of us, however, has ever seen a ghost-free 4.25 mHz picture, so we have no idea how good the system could be.

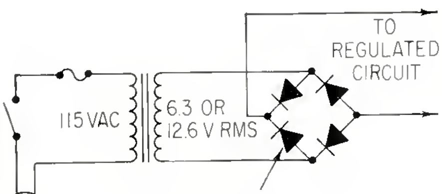
Perhaps, then, he is making a strong case for improving both picture and sound! Surely he is correct in saying that the two must remain compatible.

Ross H. Snyder
Woodside, California

The Editor:
In the March 1969 issue of your fine publication on page 28, Figure 6, in the article by Robert C. Ehle there appears a schematic that I am sure is a misprint. The schematic shows the 6.3 and 12.6 regulated power-supply circuits with the lower IN1614 diode connected backwards. During the 180-degree to 360-degree wave form of the a.c. cycle this diode will not supply the necessary pulse to the regulated section of the power supply. Please put my mind at ease and show a correction.

Richard H. Eckels
Chief Engineer
WKAN
Kankakee, Illinois

Reader Eckels is quite correct. Murphy, who gave us his noble laws, is the responsible culprit for the inverted diode. A corrected version of the supply is shown herewith. Ed.



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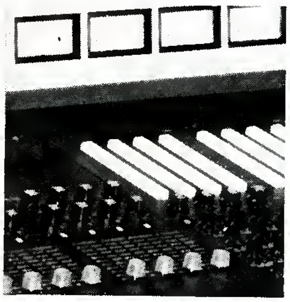

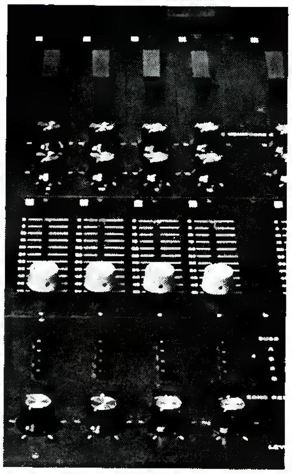

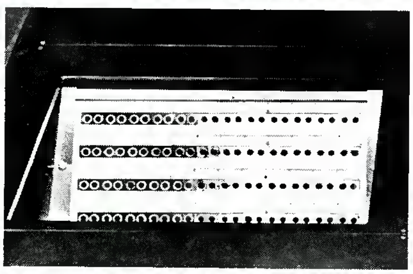

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The Audio Engineer's Handbook

GEORGE ALEXANDROVICH

MULTICHANNEL CONSOLES, DESIGN TRENDS AND APPROACHES

● Anyone who visited the NAB show in Washington D. C. and the AES show in Los Angeles was confronted with a large number of firms engaged in the construction and design of audio consoles. In the recording (and to some extent in the broadcasting) fields one can notice that the trend toward multichannel, modular, full-facility mixers is in full swing. Almost all of the recording consoles were designed to feed multitrack recorders and had the upper tier of the console filled with vu meters. Tape recorders on their part had electronics for each one of their sixteen channels each with its own vu meter. Control areas of the consoles were filled with switches, pots, keys, faders, equalizers, and lights. It wasn't hard for one to imagine that being at the controls of such a console is like flying a jet plane. It's clear that the trend in console design is one of providing all the flexibility that imagination and money can provide. It is partially understandable, customers today don't want to be restricted by the limitations of the studio equipment. Many recording groups bring their own gear—their own amplifiers, gadgets, fuzz boxes, and special effect devices. The studio has to accommodate them all.

With all this flexibility and the number of active components in the system, maintenance problems become proportional to the quantity of individual components. It thus becomes highly desirable to reduce the down time of the system from hours to minutes and seconds, by correcting any malfunctions by the expedient of unplugging the defective module and replacing it with a standby spare. It should be done without using technical personnel. Still the demand on performance of today's equipment is much stronger than ever before. With improved reliability the customer seeks performance approaching the theoretical limit simply because he knows he can get it. No longer are the upper and lower limits on frequency response 30 Hz and 15 kHz; now it is not unusual to measure response within one dB from 10 Hz to 100 kHz.

Distortion of the console is considered high, if the measured figure at any frequency exceeds 0.5 per cent and the noise figure marginal, with a noise level 60 dB below the line-level

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output. Operational flexibility of the console is equally important. Console manufacturers compete among themselves by offering features in their systems they think are the most important. If you were to collect all of the features offered by the majority of the console manufacturers and incorporate them all into one system, the description would read something like this.

Audio mixing multichannel console. So many inputs, so many outputs. Every input channel has low noise preamp, transformer input. Input connectors for all mic inputs are prewired for d.c. bias voltage required by condenser microphones. Bias voltage is supplied by the common power supply. Gain of the mic preamp is continuously variable or adjustable in steps to accommodate variety of microphone levels. (pads although extensively used are due to be phased out in the near future as an inperfect means of reducing the gain of the input stage). Feedback control of the mic preamplifier is brought out as a console control to balance the outputs of preamplifiers. Small monitor meters on modules measure the outputs of the preamplifiers detecting any excessive levels. Lights are also available instead of meters as overload detectors of the first stage. High-level input parallel to the mic input is fed into the hi-lo selector switch which followed by the fader. The fader may be a T pad, L pad or modified H. wirewound or light sensitive with remote control capability without any mechanical contacts. Cue switch on the fader may serve as a cue or prehear switch or as contacts to turn on a tally light at the microphone to indicate that it is on. Input and output terminals of the fader are fed into the echo selector switch for selection of pre- or post-echo feed for special effects. Gain control which follows the echo switch adjusts the amount of signal being diverted into the special effect or echo channel. Dry signal follows into equalizer, compressor or voice-actuated gate. Positions of the compressor and the equalizer may be changed to be before the fader if signal has to be processed prior to sending it into the special effect feed. Channel selector switch may be of the type allowing connection of the input channel output to more than one channel at the time, in effect splitting the signal between the two or more channels, creating the phantom channel during the re-recording session. A new breed of mixing networks, designed around operational amplifiers, capable of no-loss mixing of any number of channels without the need to compensate for level changes, simplifies design and construction of the mixing networks.

A unique way of channel selection using logic circuits eliminates banks of push-button switches. In the construction of the channel selection indicator, digital readout is used above each input. Using reed switches instead of manually-actuated types increases the reliability and life of the switching matrices.

Outputs of the mixing networks (amplifiers) go through the submaster faders with echo-return feed applied to the input and output of the attenuator through the post-pre echo return selector switch. Post position may be connected after the master fader control for special effects.

New types of multichannel faders allow the fading of all channels simultaneously. Output amplifiers with low output impedance supply the signal for the recorder inputs, vu meter, monitor circuit (and in some instances) for the earphones.

New solid-state vu meters provide the means of level readout that is not dependent on mechanical movements. Capable of peak reading, they also can be set for various degrees of damping, resembling the D'Arsonval movement. Also, they make more compact consoles mechanically possible, especially in multitrack systems where the dimensions of the vu meters determine the width of the board.

The extensive use of operational-amplifier circuits with their high input and low output impedances has altered the design philosophy of consoles. Several amplifiers can bridge off one low-impedance source while one amplifier can drive several lines, all without the slightest effect on the output level, distortion or noise.

Latest console designs (similar to older types) have full monitoring facilities with one difference. The number of channels monitored is usually restricted usually to four with a provision to monitor simultaneously in stereo and in mono. Monitor-select switches with position 4, 2, and mono are fast coming into being. For the purposes of prehearing each input individually solo buttons at the output of each input channel mute signals from all other channels except their own.

Talk back and slating circuits are now wired to feed preselected channels while the facility to feed earphones in the studio from separate mixer outputs can be found in almost every console. Built-in tone generators are used to check the circuits as well as for the slating.

Patching facilities although used are in the console with an entirely different purpose. While at one time patching was used to have an access to every part of the console in case of

troubleshooting, now it is used merely to add flexibility to the input and output circuits of the console. Maintenance is simplified today by modular construction so it is more economical to provide additional channels and spare modules than to use the patchfield to gorrow the facility from an unused channel.

All of these innovations are representing progress in the right direction. Sometimes, though, they are part of systems that have been overdesigned in an effort to impress the customer. Mixers that are overly complicated and cluttered with an excess of flexibility are comparatively easy to use during the original sessions but require real talent to conduct a mixdown session with good results.

It is a welcome sight to see so many useful innovations in the audio mixer field especially when they bring improved performance, simplified circuits, and reliability. Extensive use of solid-state devices cuts the number of mechanical devices to the essential minimum.

The solid-state vu meter is one of them, especially since new circuit eliminates loading of the circuit and the distortion associated with rectifier type movements.

There are a number of things one wishes would be developed as you read about the possible applications of the modern technology. Wouldn't it be perfect to have (instead of ordinary vu-meter scale) a screen of the same size, horizontally graduated in hertz and vertically in dB. Then as the signal would be applied to the device a family of rising vertical columns would appear representing the frequency and the amplitude of the signals. A bright red line, three quarters of the way up drawn horizontally across the screen, would indicate the maximum safe mixing level.

Or there could be more tangible things such as phase-indicating devices, be they scopes, lights, or other types of indicators? Do we really know the right phasing of our output signals? And when we find out, what can we do about it? Why not design a "phase rotator" so that when the out-of-phase condition is detected that it can be corrected. Each mic channel could use one. Why not a built-in limiter for the first preamplifier to eliminate the overload of the entire system?

Isn't technology ready to give us the ability to design a mixing board for 16 inputs and 8 outputs the size of a two-by three-foot picture frame or smaller and one-inch deep, for remote work? After all, amplifiers for such a console could be fitted into a space smaller than a pack of cigarettes with most of the space representing packaging and interconnections. Transistor wafers are so small that they can hardly be seen with the naked eye. We have learned to eliminate most of the bulky components such as transformers, tubes, chokes, large electrolytics and other components, from our circuits.

Packaging techniques have advanced to a point when it is possible to assemble close to *one million components per cubic inch.*

Well, I think the time has come for all of us to realize the importance of semiconductor technology along with its impact on our way of approaching the design problems. We have come to realize that battleship type of construction is not always the best, be it from the standpoint of portability, performance, maintenance cost, or operation. We have learned to make use of new components for better systems, new technology to simplify design, and new circuits for better performance. We have to do the same with our techniques of recording—bring them up to date and let us stampout notions that believe that running vu meter pointers into stop cushions will make music sound better.

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

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

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

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Neve designed and built it for **Vanguard** Records

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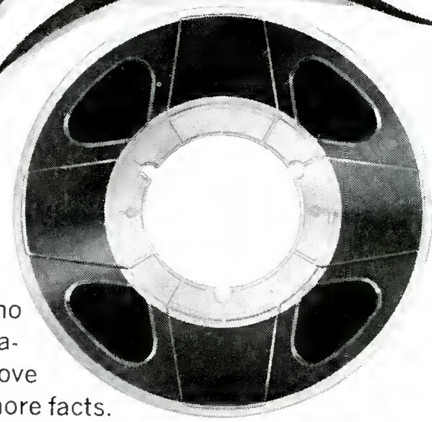

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The Feedback Loop

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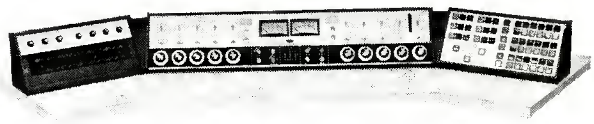

Figure 1. The console module used in the control room at KMOX.

●What's doing at CBS Radio?

I recently had the opportunity of talking to Ogden Presthold, Director of Engineering CBS Radio. Oggie is an intelligent and knowledgeable engineer doing an important job for the CBS Radio Network. His office is located in the CBS building at Avenue of Americas and 52nd Street in New York City. (After twenty-five years with CBS, Oggie will be leaving this summer to become a partner at A. D. Ring, engineering consultants. His replacement will be Ralph E. Green, presently director of operations at WCBS.) At CBS Oggie and his staff have, among their other duties, the job of designing and supervising the construction of network facilities, as well as owned and operated local radio stations. CBS' radio engineering staff gets a lot more practice at designing control rooms and studio facilities than the engineering staff of an independent local station. For this reason, I think, there's a lot to learn from Oggie's experience and from the approach that he and his staff have evolved.

Central to the current CBS approach is the concept of modular construction. Consoles, turntables, tape recorders, and other modular elements that make up a control room are fabricated for plug-in installation. The control room can be built up to its required size or complexity by the addition of standard plug-in modules. The modular approach, Oggie feels, when applied to the individual reel-to-reel tape recorder converts that device to a small control area in itself; incoming programs can be recorded, taped programs can be transmitted to other areas, or fed directly to the outgoing program line, and finally the tape recorder can function as an editing and production facility. By having all the necessary feeds brought directly to it and available on a selector switch, the tape recorder can be operated independently with maximum utility and efficiency. The essential features of the modular approach are to have interchangeable elements in use throughout an installation, with connectors used to install the equipment.

The CBS approach to preselection of the numerous feeds, which must be available at all console and tape recorder inputs, is somewhat different than that used in the ABC Radio Network facility (discussed in the February and March Feedback Loop). At CBS, program lines are brought directly to selector switches. Switches are used wherever a selection of feeds is available. The use of jack fields is avoided wherever possible because they represent a possible source of trouble in that the line may be shorted or opened.

A good example of the CBS approach

Figure 2. The turntable module used in the control room at KMOX. Note the cue amplifier below the right turntable.

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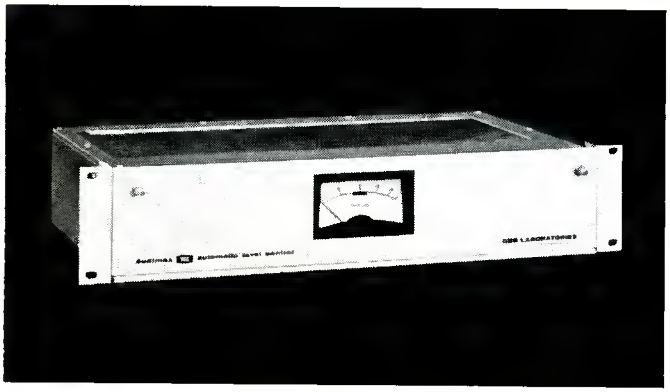

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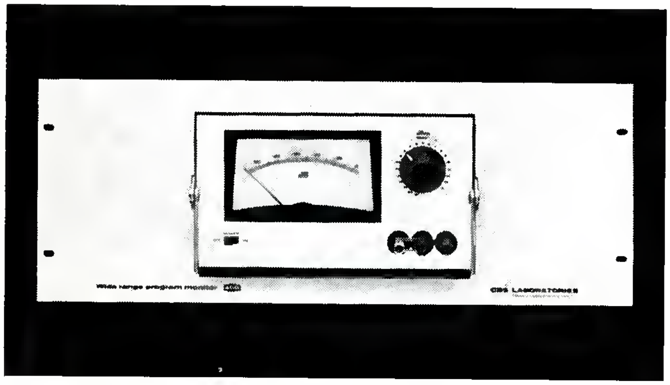

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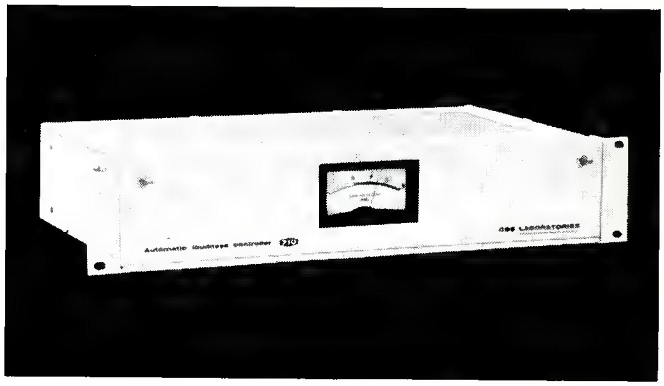

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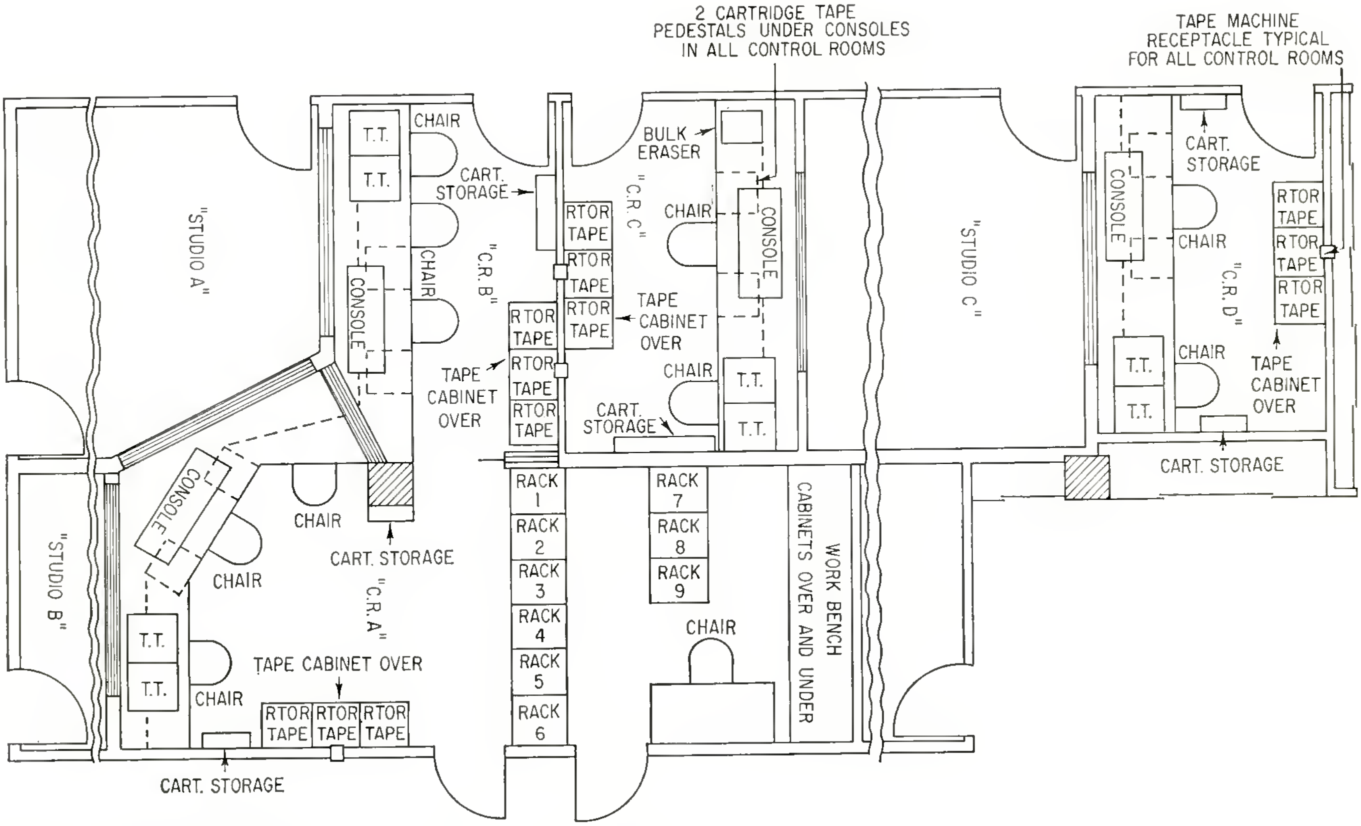


Figure 3. The Station KMOX floor plan. Space requirements have forced us to compress the studio space shown at both ends.

in practice is the recent KMOX installation. Oggie described how station KMOX (located in the Gateway Tower Building, St. Louis) was conceived, designed, and constructed under the guidance of his staff. It all began with extensive discussions between the engineering staff, program people, management, and the news department. In a large organization like CBS, the group which designs and constructs KMOX is separate and distinct from the group that is now operating it. The engineering staff must have an intimate understanding of the operational philosophy of the station and an awareness of current operating problems. This makes the initial discussion stage extremely important. Working with this concept the engineering staff developed a set of plans which specified the functioning and electrical characteristics of the KMOX facility. This information is assembled in a document called *Audio Equipment Specifications.* This booklet was the basis for obtaining bids from outside manufacturers for the construction and installation of KMOX. An interesting feature of the specifications is the precise and concise way in which the electrical characteristics such as frequency response, noise, and distortion specifications were stated. This information — and there is a great deal to specify in a station with five control rooms and four studios — is contained on one 8½x11-in. page. The specifica-

Figure 4. The control console photographed under test at the McCurdy factory.

tions are displayed in graphic form which is both easy to read and the information contained is immediately accessible. Graphic displays sometimes convey information more rapidly and accurately to the engineering mind than a comparable set of numbers and words.

The KMOX station layout is shown in FIGURE 3. Control Room "A" is so located that there is access to the central electronic complex — racks 1 through 6. This control room also fronts on both studios "A" and "B". The control console and turntable modules used in this control room and throughout the station are shown in FIGURES 1 and 2. To the right of the console (FIGURE 1) we see the readout and control panel of the *on air* switcher. On the left-hand side we see the rotary selector switches for selecting program feeds. Reel-to-reel tape recorders cannot be conveniently located adjacent to the control console. However, since each recorder is operated independently, with its own feeds and input selector switch, no problem is encountered in their operation.

Now comes something which was a complete surprise. Less than two weeks after the equipment was delivered to the KMOX site in St. Louis, the installation was completed and the station was on the air from the new studios. Modular plug-in equipment and prefabricated cabling are the secret. The equipment and cables (all cables terminating in appropriate connectors) were fabricated at the McCurdy Radio Plant, McCurdy being the successful bidder for the KMOX installation. Ductwork was made to accommodate both cables and connectors. The entire system — equipment, cabling, and connectors — was thoroughly checked out at the factory, and FIGURE 4 shows a console and its prefabricated cables under tests at the McCurdy Plant. This procedure minimized the problems encountered at the installation site after the equipment was put into operation. Upon delivery the modules and prefabricated cables cables were plugged in and the installation was then complete and ready to go on the air.

KMOX is a very interesting and innovative approach to radio-station installation, representing a great deal of engineering talent and planning.

Theory and Practice

NORMAN H. CROWHURST

●Converting audio signals into acoustic waves is one of the most discussed problems of our field. High efficiency, low efficiency? One unit, many units? And all the variations. If multi-way systems are used, how sharp a crossover should be used? And should it be before or after power amplification?

The problem with having so many variables is that one must make certain assumptions, valid or not, before one can make comparisons of even part of the problem. If we generalize about whether or not to use multi-way systems, we must assume some sort of ideal frequency division system — and that's a problem in itself. On the other hand, if we assume we are going to use a multi-way system somehow, and seek the ideal frequency division arrangement, we are committing ourselves by implication to the notion that this is the best way to do it, when it may not be.

For the moment, let's leave the question of multi-way or not to the acousticians, and consider some purely electrical — or electronic — factors connected with frequency division. First, how sharp?

Now we have to come back into the question of speaker performance, like it or not. For our prime requirement is the best over-all result. If our speaker units work well only within their specific ranges, and "foul things up" outside their own ranges, then frequency division needs to be quite definite. On the other hand, if their deterioration outside their own range is quite gradual, a more gradual crossover transition is quite acceptable, in fact preferable.

Many multi-way systems use horn-type units, for at least some of the frequency ranges. These units are characterized by quite sudden cut-offs, often at both ends of the range. So crossovers for use with multi-way systems that use any horns at all need to be quite sharp in their transition — at least 12 dB per octave. On the other hand, a system that uses all cone units, suitably mounted, will not have this problem, and 6 dB per octave may be quite adequate.

That's one angle. If you only think about that, it seems you've got the picture straight. Now you only have to decide which combinations of units do the job best, and select an appropriate crossover system. But that's not all to the how-sharp question.

The prime objective in putting together a system is to handle all segments of the frequency response as smoothly as possible. And the over-all response should be as flat as possible. And so it may be, measured with a steady-state signal, or a slowly gliding tone — or even a warble tone — to eliminate standing waves from the measurement.

But what about transients? If you use sharp crossovers, to suit speaker units that need that variety, they are designed to give constant total output over the frequency range. But in another sense (the transient sense) this means you have put in some built-in peaking.

If you cascade 4 r-c networks to produce an ultimate 24 dB/octave cut-off each way, with 180-deg. phase shift in each network at crossover, and no interaction such as happens in a resonant circuit, your response is 12 dB down in each output at crossover. Thus the combined power is 9 dB down, which is quite a dip.

So, however you do it, whether by using L's and C's in an electrical crossover, feeding the units themselves directly, or whether you use electronic crossovers before the power amplifiers (FIGURE 1), the only way you can get an over-all, combined response without the dip, in effect puts a peak into each speaker circuit (FIGURE 2). On steady-state testing, this peak can be viewed as "holding up" the loss, until you're ready to "let it go."

But to transients, a peak is a peak. It will ring. The sharper the crossover, and however it is put into the circuit, the more it will make transients "twang" in the vicinity of crossover frequency. So much for the how-sharp question; now we'll turn to where to put it.

First, whether you use an electrical crossover feeding the speakers, or an electronic crossover that produces the same response by employing feedback with r-c networks, the relationship between phase and amplitude (dB) response is the same, when each is done correctly for the end result to be the same.

So the question of where to put it should be related, not to frequency response or transient performance, but to distortion aspects. One of the reasons for using separate units is to keep possible sources of distortion separate, and thus minimize intermodulation distortion of various kinds.

While this might not seem so important in relation to amplifiers as it is for loudspeaker units, here's a point about amplifiers that's often overlooked. To illustrate it, we'll have to assume some figures related to program level in the different frequency ranges.

Let's assume that the program is to be fed to a three-way system, and the output waveform reaching each may be specified as follows:

Low-frequency unit: waveform representing 64 watts peak (requiring an amplifier capable of delivering 32 watts rated), and having average power content of 10 watts.

Middle range unit: peak power 50 watts (requiring 25-watt rated amplifier); average power content 8 watts.

Tweeter: peak power 25 watts (requiring 12.5-watt rated amplifier); average power content 4 watts.

If you use separate power amplifiers, with electronic separation before them, you can handle this signal comfortably with amplifiers rated at 35 watts, 25 watts and 15 watts, adding up to a total power rating of 75 watts.

So wouldn't a 75-watt amplifier, with electrical crossovers to feed the correct frequency range to each unit, do the same thing?

Waveforms add voltages or current, not watts. So let's see what voltages are involved. The low-frequency unit, assuming 16-ohm units throughout, needs to handle 32 volts peak, 12.6 volts r.m.s. The mid-range needs to handle 28 volts peak, 11.3 volts r.m.s. The tweeter should get 20 volts peak, 8 volts r.m.s.

The r.m.s. values don't add. You have to take the squares, add and take the square root, to get total average power. Or you could do it direct, by adding the average power contents, 10 + 8 + 4 = 22 watts, which corresponds with 18.8 volts r.m.s.

But as all the waveforms have different frequency content, there are times, a few times every second, when all the peaks add, although the average is more

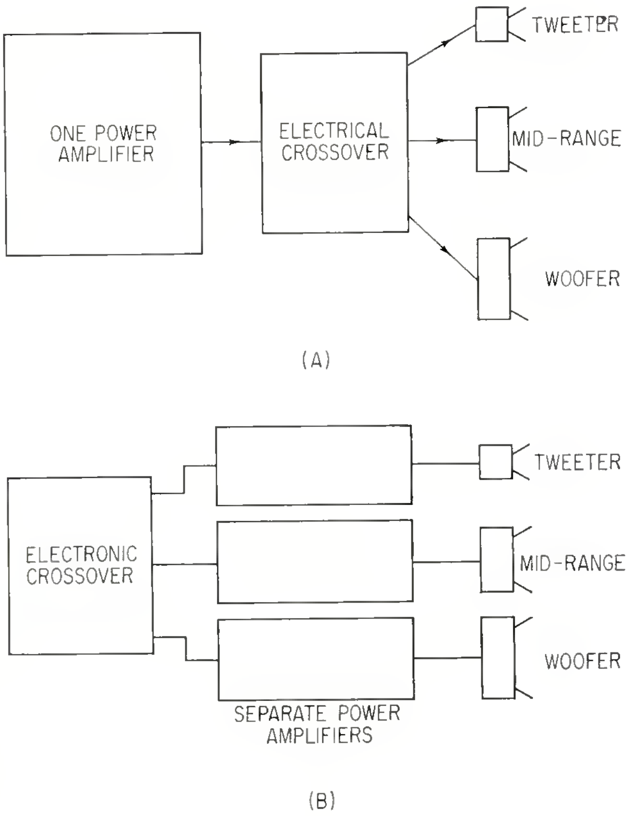


Figure 1. The difference between an electrical crossover (A) and an electronic crossover (B).

like the r.m.s. figure. And the system must handle the peaks, to avoid distortion. So a single amplifier to deliver power to all three units, with an electrical crossover, needs to handle the sum of the peak voltages $32 + 28 + 20 = 80$ volts peak.

This represents a peak power of 400 watts, which would need an amplifier rated at 200 watts — considerably in excess of the 75 watt total rating, using separate amplifiers.

So far, although we didn't go into that aspect in detail, our use of more than one unit, with the notion that two units may be better than one, and three may be better than two, has concentrated on using each unit for the frequencies it handles best, and thus combining essentially different units. That is what got us into frequency separation.

And of course, the electrical-watts requirement is also dependent on how efficient, or more precisely how inefficient, the units are. Thus, if we use units with a conversion efficiency of 20 per cent (which is fairly high for a loudspeaker) in place of on that is 2 per cent efficient, we shall get the same amount of sound for one-tenth the electrical power.

If those wattages calculated to illustrate frequency division were based on using units that are 2 per cent efficient, and we could trade them for units that are 20 per cent efficient, a 20-watt amplilifier would serve all 3 units as well as a 200 watt amplifier would do with the 2 per cent efficient units. So efficiency is more important in the over-all question than immediately appears.

But why use essentially different units for different frequency ranges? The concept is that big speakers handle low frequencies more efficiently and small speakers handle higher frequencies better. However, they don't have to be different. A number of smaller speakers can handle the lower frequencies as effectively as one larger one, and the smaller ones still handle the higher frequencies just as well.

More than that. Mounting a number of small speakers on a common baffle makes them operate more efficiently than working them individually. So this may well be the better approach. The reason for the improved efficiency should be explained.

Any speaker unit has an essential mismatch, acoustically, between the diaphragm and its driver mechanism and the air into which the sound waves are propagated. This is oue factor that limits the ultimate efficiency of speakers, so that a figure like 20 per ecnt is considered high.

The diaphragm and its driving mechanism is inevitably mnch heavier than the air it drives, consequently it expends more of its energy to drive itself, than it does to push the air to make sound waves.

When a number of speakers are connected together, so they all do part of the wave pushing, each speaker creates an increased back pressure in the air for its neighbors to work into. This means each speaker works as if the air is heavier than it seems when each works individually, and thus more of energy gets used in moving air, and less in moving itself.

A way of observing this is to compare identical units mounted singly and in groups, on baffles of identical size (but with holes to suit one or many speakers). The small speaker in a large baffle has to move its diaphragm quite visibly to produce appreciable radiated sound volume. The many small speakers working in unison produce a greater radiated sound volume, while visual inspection of their diaphragm movement shows that any one of them is doing very little work to produce the stronger sound wave.

So what at first appears like a "cheap" way of doing the job — by mounting a lot of inexpensive speaker units on a fairly large common baffle — may really be a superior way of doing it, after all. Perhaps I should conclude this month's discussion by emphasizing that I certainly will not infer that one way is inherently better than any other way: the problems are different, according to how the job is tackled. The best speaker is the one that makes best use of its own method.

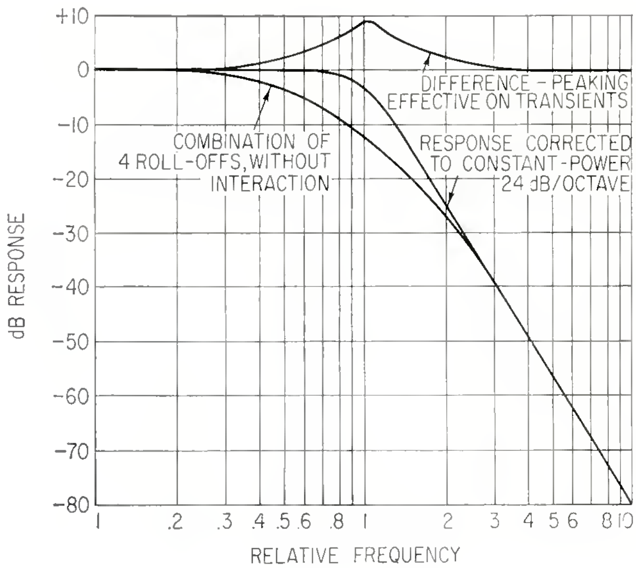
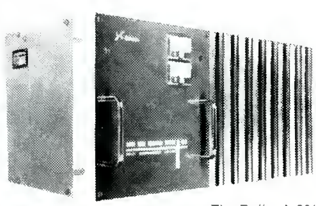


Figure 2. This is how achieving the correct steady-state frequency response results in peaking effective when handling transients.

MOVING?

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



The Dolby A 301

Sound with Images

MARTIN DICKSTEIN

APPLICATIONS OF CLOSED CIRCUIT TV

● In the last twenty years, the field of closed circuit TV has moved rapidly from a new-born idea, through infancy, and into adulthood. But it is still growing and will probably not reach full maturity for quite some time to come.

Each industry, in its own way, has has a hand in the growth of the CCTV business. Most applications are fairly simple for a CCTV application. In a parking lot, cameras are located (with or without remote controls such as pan-and-tilt, zoom lens, heater and windshield wiper on the weather housing, etc.) either on poles or on the side of the building facing the lot. Using a constant-focus lens with pan-and-tilt unit or several cameras to cover the same area will depend, of course, on the locations chosen for the mounting and the lens on the camera. Night-time coverage will require special lighting in the areas to be covered. For applications such as this, outdoor housings were developed with various assessories to combat the effects of weather.

Another fairly obvious application for CCTV is traffic control. Here, again, the installation of the fairly delicate cameras has been outdoors, and the outdoor all-weather housing is used for protection. Choice of location for the cameras requires careful surveys to determine coverage angles, cable runs, traffic directions and peaks, etc. One innovation incorporated in this use has been the V-shaped mirror in front of the lens of each camera required to look up and down a street or tunnel. The resulting image in the monitor is a split-screen effect with the observer able to see each way on one thoroughfare.

Many specifications for CCTV installations describe the intent of the system as "extending human vision". This indicates the possibilities of putting a camera where a human can not be or cannot stay for any length of time, such as in front of a furnace in a power plant, near reactors, between machines in a factory production line, on top of a crane in a construction project or in a railroad yard, on top of a bridge tower either watching motor traffic or overlooking a river entrance, in a large storage area or on a railroad or subway station. Surveillance of building doorways, driveways, lobbies, basements, bank vaults, department stores and tight-security locations are naturals for a carefully hidden or very obvious camera (depending on the customer's viewpoint).

Most recently, of course, the use of a camera in outer space has been tried with great success, a real treat for the earth-bound human whose vision was extended to where most of us will never go.

A tremendous field has sprung up in the last few years in educational CCTV. From junior high schools to universities, closed circuit is being used in a tremendous variety of ways. The simplest situation is pointing a camera at an instructor or lecturer in one room and having classes in other rooms able to see and hear the demonstration. The system is usually enlarged to include the auditorium in many schools. Large monitors, strategically placed, can cover a very large audience. The advantage to a system such as this is that every person in the audience can actually look at the hands of the lecturer as the demonstration progresses even if the observer is in the last room down the hall across the city.

Educational television has now advanced to the point where the student can be home or in a hospital and still not miss out on the daily lessons. Two-way systems keep the patient right in the front row of the class room.

In industry, CCTV has found use in the training classrooms, in conferences between people in different buildings located in far-apart spots within or outside of the city. In New York, a large bank with offices in different parts of the city has a CCTV link between them to expedite meetings without wasting the time it would take to travel from one building to the other. Executives find it most convenient and relatively inexpensive (over a long period of time) to be able to "meet" face-to-face and even show each other documents without having to leave their own buildings. Banks have closed-circuit systems between the tellers and the back offices for signature verification, for transactions through a window without either the subscriber or the bank employee being anywhere near each other and most recently for monitoring teller trainees while they practice handling money and paperwork while dealing with "customers".

Hospitals have also jumped on the bandwagon and now have systems between operating rooms and pathology, between operating rooms and intern classrooms and lounges, between X-ray and the resident's office, etc. CCTV is also used for remote observation during Auoroscopy and, of course, between the floor nurse and the intensive care or other patients. The system also includes the mounting of a small TV receiver on a boom-like arm near the patient's bed for regular broadcasts with a nurse's override whenever it is necessary to communicate with the patient. In fact, these systems are now being wired into new hospitals with only one coax cable between the patient and the head nurse. This single cable carries all channels of transmission, voice intercom, the nurse's face and any other pertinent information desired.

It has been found, also, that a patient is made more comfortable if they learn a bit of the operation of the hospital. The CCTV system includes on a local channel, information given by one of the administrators regarding operations of cleaning up, running the business office, different devices used in treatment, etc. Some hospitals even provide the patients with special programs put on by members of the hospital staff, other patients or outside talent such as school groups or even visiting professional personalities.

Now, add to all the above ingredients a new gadget — the video tape recorder. Each system can now expand its use of the CCTV equipment. In banks, the tape device is started to record the actions of suspicious individuals and no time is lost in playing back the tape when required. In factory production lines, a record is kept of an operation and this can be analyzed at a future time for either evaluation or expansion and retooling. Space recordings, of course, are invaluable. But the greatest application, outside of broadcasting, has become the school or university customer. Of course, the training of executive personnel, or the possibility of an executive of rehearsing his talk so that he can improve it before going in front of the stockholders or the board of directors, is another application in industrial use, but the instructional field benefits the most. Demonstrations, lectures by famous professors, courses recorded by an instructor for playback at any time to relieve the teacher of the necessity of having to do the experiment over each time a new class comes to the course — the uses are limitless.

And, since the introduction of the video tape recorder, the field of psychology has come into the long list of CCTV users. Prior to the point where the reactions of the patient and psychiatrist's questions could be put on tape, reports had to be made verbally by the doctor or the researcher. Now, it became possible for the doctor to review at a later time, the reactions of the patient. A hand movement here, or a blink there or a twitch which the psychiatrist missed during the interview can now be seen. The facial reaction of the patient can be studied and is not lost forever after the discussion. Psychiatrist students, after having been told about certain patient actions, can now see in detail what they had been told.

The newest introductions to the field of CCTV are the methods of editing tape electronically by using a computer-type control unit to switch from the A reel to the B reel at a precise point without a razor blade being lifted, and the electronic video recording process whereby a picture is placed on film and played from a cartridge. This device in particular seems destined at present to invade the educational field.

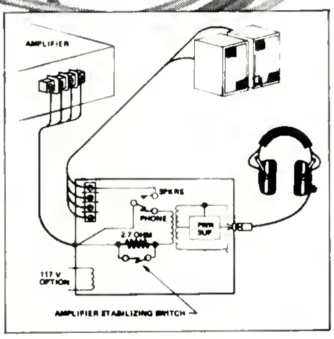
It seems as though the sound contractor will have to get involved in CCTV in spite of anything he can think of to the negative. The field of closed circuit is in every phase of everyday islife and will soon be in the home. CCTV is still growing, it is not too late to get into the field. Even though it would not be on the ground floor, at least it will be before the speeding elevator reaches the penthouse.

model esp-9 studio monitor

Offers a new tool for improved audiometry with self-bias or ac line energizing. Controlled monitoring with 10 octaves ± 2 db range exceeding best loudspeaker systems. Eliminates all room reflections and polar distribution problems.

Rear view of the E-9 Energizer.

Functional Block Schematic showing switching functions and hook-up of the ESP-9.



NEW!

SUPER, WIDE-RANGE RESPONSE for critical, controlled monitoring of finest recording sources. Delivers all 10 audible octaves, 15-15,000 Hz ± 2 db, 4 octaves beyond ordinary headphones.

VIRTUALLY DISTORTION-FREE PERFORMANCE through precision electrical balancing of push-pull acoustical circuitry to give fatigue-free listening through long, intense recording sessions. Elements cancel all 2nd harmonic distortion, unlike conventional units.

LIGHTWEIGHT—HUMAN ENGINEERED FOR COMFORT— Uses fluid-filled cushions for distributed gentle pressure with good seal; coupling transformers and circuitry located in external housing; extendable stainless steel headband with wide cushion for perfect fit and restful listening.

CALIBRATED, PRECISELY CONTROLLED OUTPUT—IDEAL FOR AUDIOMETRIC USES— Switch on front panel of energizer selects ac operation for precision measurements of output; in self-energized switch position no connection to ac lines is required; this gives maximum convenience.

HIGH-POWER CAPABILITY IN VERY LOW BASS RANGE—Large, oversize coupling transformers mounted in E-9 energizer unit give good wave form at 30 Hz with up to 10 volts input.

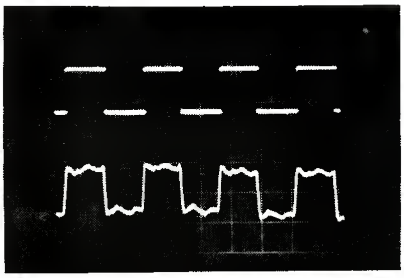
NO SPECIAL AMPLIFIERS REQUIRED—CONNECTS TO LOW-IMPEDANCE SPEAKER TERMINALS—Easy, quick hook-up to any good amplifier delivers performance to specification.

The ESP-9 is a refinement of the famous ESP-6 Electrostatic Stereophones. The most important new feature is a response range of 10 octaves, **the widest ever attained in a headset.** A new cup design promotes virtually linear response to below 20 Hz.

The ESP-9 has a signal handling capacity of 10 volts at 30 Hz with good wave form versus 6 volts for the ESP-6. This is made possible by increasing the size of the coupling transformers by a factor of 4, and mounting them externally to the cup in the E-9 Energizer.

The E-9 Energizer offers the option of self-energizing for the bias supply, or energizing through the ac line; choice is made with a selector switch on the front panel. When energized through the ac line, very precise level measurements can be made. Thus the unit is ideal for audiometry, and for evaluating the spectral character of very low level noise in equipment like tape mastering machines and recording consoles. In contrast to the ESP-6 and ESP-7, both cups are independently energized; a left cup signal is not required to supply bias to the right cup.

TYPICAL SQUARE WAVE RESPONSE AT 400 Hz.
Trace at top is input, lower trace is ESP-9; note unusually close resemblance.



ELECTRICAL SPECIFICATIONS

Frequency Response Range, Typical: 15-15,000 Hz ± 2 db (10 octaves) 10-19,000 Hz ± 5 db. An individual, machine-run calibration curve accompanies each headset. This curve uses standard 3-1/2 log-cycle chart paper, and reads from 20 to 20,000 Hz only.

Sensitivity: 90 db SPL at 1kHz ± 1 db referred to 0.0002 dynes/cm^2 with 1 volt at the input. Variations from calibration furnished are less than 1/2 db at 25°C.

Total Harmonic Distortion: Less than 1/5 of 1% at 110 db SPL.

Isolation From External Noise: 40 db average through fluid-filled cushions provided as an integral part of the headset.

Power Handling Capability: Maximum continuous program material should not exceed 10 volts (12 watts) as read by an ac VTVM (Ballantine meter 310B or equal) with average indicating circuitry and rms calibrated scale; provides for transient peaks 14 db beyond the continuous level of 10 volts.

Source Impedance: Designed to work from 4-16 ohm amplifier outputs. At higher impedances response at the extremes of the frequency range will progressively reduce; e.g., 50 ohms causes a loss of 5 db at 30 and 10,000 Hz.

External Power Requirements: None, except when used for precise low level signal measurement, when external ac line can be selected by a front panel switch on the E-9 Energizer (1/16 amp, 117 VAC, 50-60 Hz normally; 234 VAC with internal strap for foreign use).

PHYSICAL SPECIFICATIONS

Size of Cup: 4-1/4" h x 3-3/4" w x 1-1/4" d.

Cushions: Fluid filled for high ambient noise isolation.

Headband: Extendable, stainless steel bands with self-adjusting pivoting yokes; conforms to any head size.

Headband Cover: Formed of wide, soft molded-rubber with 1/2" polyethylene sponge cushion on underside.

Boom Mount for Microphone: Knurled, anodized, aluminum knob on left cup with threaded shaft and 2 compressible rubber washers; accepts all standard booms.

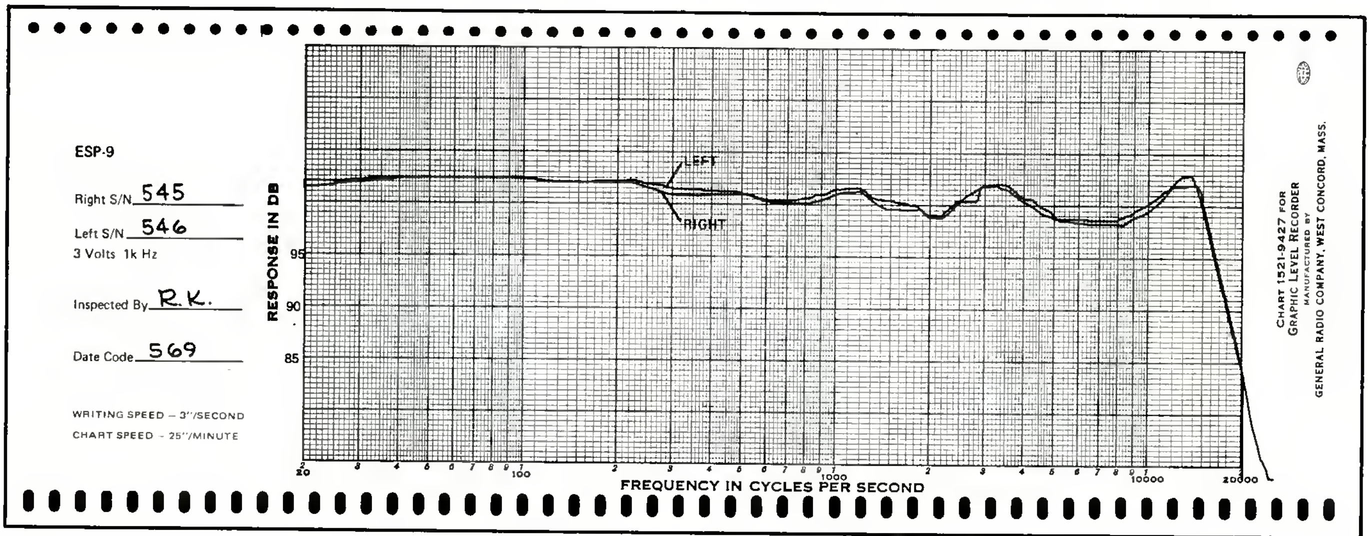
Headset Cable: Flexible, polyvinyl, 5 conductor, shielded, 6' long, black, with 5 prong plug keyed to E-9 Energizer receptacle.

Weight of Headset Only: 19 ounces

E-9 Energizer: Contains 2 coupling transformers, self-energizing circuitry, speaker/headphone transfer key-switch and ac pilot light on black anodized front panel. Also contains ac power transformer, ac on-off switch, ac line fuse, and speaker terminals. Size is 4-1/2" h x 3-3/4" w x 6-1/4" d; weight 3 pounds. Has 6'4 conductor input cable terminated with 4 spade lugs to connect to amplifier output terminals.

Accessory Provided: 6' ac line-cord P/N 41-0235 for optional use, with plug on one end and plug-receptacle on the other.

Model ESP-9 Studio Monitor: Electrostatic Stereophones, complete with E-9 Energizer, ac line-cord, machine-run calibrated response curve and instructions; Shipping weight 6 pounds; Price **$150.00**



MACHINE RUN RESPONSE CURVE OF THE MODEL ESP-9

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Circle 73 on Reader Service Card

Editorial

MORE ON MULTI-TRACK RECORDING

GEORGE ALEXANDROVICH'S recent columns on multi-track recording have stirred up a great deal of reader interest. As a technological breakthrough, multi-track recording already has had a profound effect throughout the industry. Now it is up to the creative audio pro to make the best use of its potential.

At a recent visit to a recording studio, I had occasion to listen to several eight-track masters of a big jazz orchestra. Each of the instrumental groups had been close-microphoned. The resultant listening experience, in six-track monitoring, two-track stereo mixdown and mono mix, showed clearly what was gained with multi-track recording. . .as well as how this exciting technique can be misused.

In all cases, the sound was absolutely sumptuous. With proper use of the control console, a fine stereo and mono mixdown (replete with judicious amounts of added equalization and reverb in the correct places) was achieved. The stereo mix had a good lateral spread and specific placement of instruments. The mono mix properly combined all the elements into a single channel.

But something was missing. In a normal big-orchestra setup, musicians sit in groups clustered alongside of and behind other groups. The particular microphone techniques used in these samples put all the musicians on a close-mic plane. The listening result was that the musicians seem to be spread over a single lateral plane (in stereo) or all occupy the same exact place (in mono). The tapes had no feeling of spatial depth.

This, perhaps, is a typical example of multi-track recording gone wrong. But surely the multi-track technique was not at fault. Rather it was the studio that handled it poorly. Multi-track recording will not destroy depth perspective if the engineers and producers consider its preservation at the time the session is being set up.

George Alexandrovich raised questions about the repeated use of the same tape and the individual condition of certain tracks on the wider tapes used today. These purely technical problems will be answered in time; they do not detract from the opportunities of the system.

Basically, multi-track is a creative new tool designed to serve producers and engineers. Only long-range experimentation will fully realize its immense potential. Today's multi-track techniques already have vastly enhanced the quality of records. As for the future, we see multi track stimulating the growth of the entire recording industry as well as suggesting many innovations still undreamed of in audio.

L.Z.

The Ultimate Noise

MELVIN C. SPRINKLE

The author explores the mathematics and practicalities of electronic noise that still lurks below the surface in much of our equipment. The understanding of its properties as presented here can do much to push it to its ultimate.

IT'S A NOISY WORLD. Those of us who live in metropolitan areas have our ears continually assailed by a cacophony of noises, including the nocturnal serenading of lovemaking cats as well as the rattle of garbage cans dropped from the collecting truck, the hucksters of television, the noises of motor traffic, and the roar of airplanes, to name but a few.

And those of us who are concerned with the electrical transmission of sound or its recording are very much concerned with noise, since one of the basic measures of reproduced sound quality is the signal-to-noise ratio — a term that has been around since the beginning of the industry and which goes back to before most of us were born.

Webster defines noise as ". . . sound of any sort, especially if without agreeable or musical quality." He further qualifies noise as ". . . suggesting meaningless, confused or discordant sound." While the dictionary was written by lexicographers rather than audio engineers, the meaning to audio men is quite clear: noise is any undesired signal which contaminates a desired signal.

In this paper we are restricting the term noise to the *natural* noise of the universe, sometimes called *thermal* noise. We are not concerned with such noises as power-line hum or the harmonics thereof which formerly plagued audio men. Thanks to transistors and better power supplies, the "humbug" is pretty much laid to rest in any quality or professional audio equipment, although he can be resurrected very rapidly by an unshielded wire or an open ground connection.

All audio men have at some time or other run into thermal noise problems. We have heard it as a hiss in high gain and otherwise quiet amplifiers. Interstation noise in an FM broadcast receiver is an example of such noise. Thermal noise is sometimes called *white* noise, *gaussian* noise or *Johnson* noise (the last-named term preceded the oratory of the former White House occupant). Reference 1 provides an excellent treatment of the nature of thermal noise and which may be summarized:

1. The waveform of thermal noise never repeats itself exactly; it is random in nature. 2. Thermal noise has no period and therefore if the waveform is analyzed it will be found that frequency components occur equally or are of equal magnitude across the bandwidth of the noise source; or (said another way) the power spectrum of a thermal-noise source is flat with frequency. 3. Instantaneous peaks of various heights occur. If measurements are taken over a long enough period, all magnitudes can be recorded. The distribution or frequency of occurrence of the several peak values follows a normal or gaussian distribution.

This last point brings up an interesting observation. In a simple sine wave the peak value of the voltage is related to the r.m.s. value by the factor 1.414 or 3 dB. If a thermal noise be analyzed it can be shown mathematically that peak amplitudes greater than the r.m.s. value by a factor of four occur less than 0.01 per cent of the time. Thus the peak-to-r.m.s. ratio of thermal noise is usually considered to be 4:1 or 12 dB. Thus an amplifier designed to amplify thermal noise will have a power handling capability 9 dB less than when

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Melvin C. Sprinkle is a project engineer with Page Communications Engineers, Inc. of Washington, D.C.

handling sine-wave signals. It will be noted that this figure is quite close to the usual 10 dB "headroom" which is the practice in well-designed audio systems.

The term *thermal* noise is perhaps an apt one, since physicists tell us that the electrons in the various atoms of a conductor are in continuous random motion at temperatures above absolute zero (minus 273°C) or 0°K, and their activity increases with temperature. Since an electron is a negative charge of electricity, the motion of the electrons produce a random voltage across the ends of a conductor and whose average value is zero. This random voltage is then the source of the thermal noise, or the ultimate noise of the universe.

Papers by Johnson,[2] Nyquist,[3] Llewellyn,[4] and Williams[5] in the late 1920's and early 1930's established quantitative values for thermal noise. The voltage developed by thermal noise is given by:[6]

$$E_2 = 4KT\int_{f_1}^{f_2} R_{df}$$

where k- Boltzman's constant = 1.374×10^{-23} joule per °K

T - Absolute temperature in °K

R - Resistive component of the impedance across which the noise is developed

f - frequency in Hertz

For practical calculations, especially those in which the resistive component is constant across the band of interest, the following expression is used:

$$E^2 = 4kTR\,(f_2 - f_1) \qquad (2)$$

The term $(f_2 - f_1)$ really means the noise bandwidth as defined by Friis[7] in his classic paper, but for most calculations the following form is used:

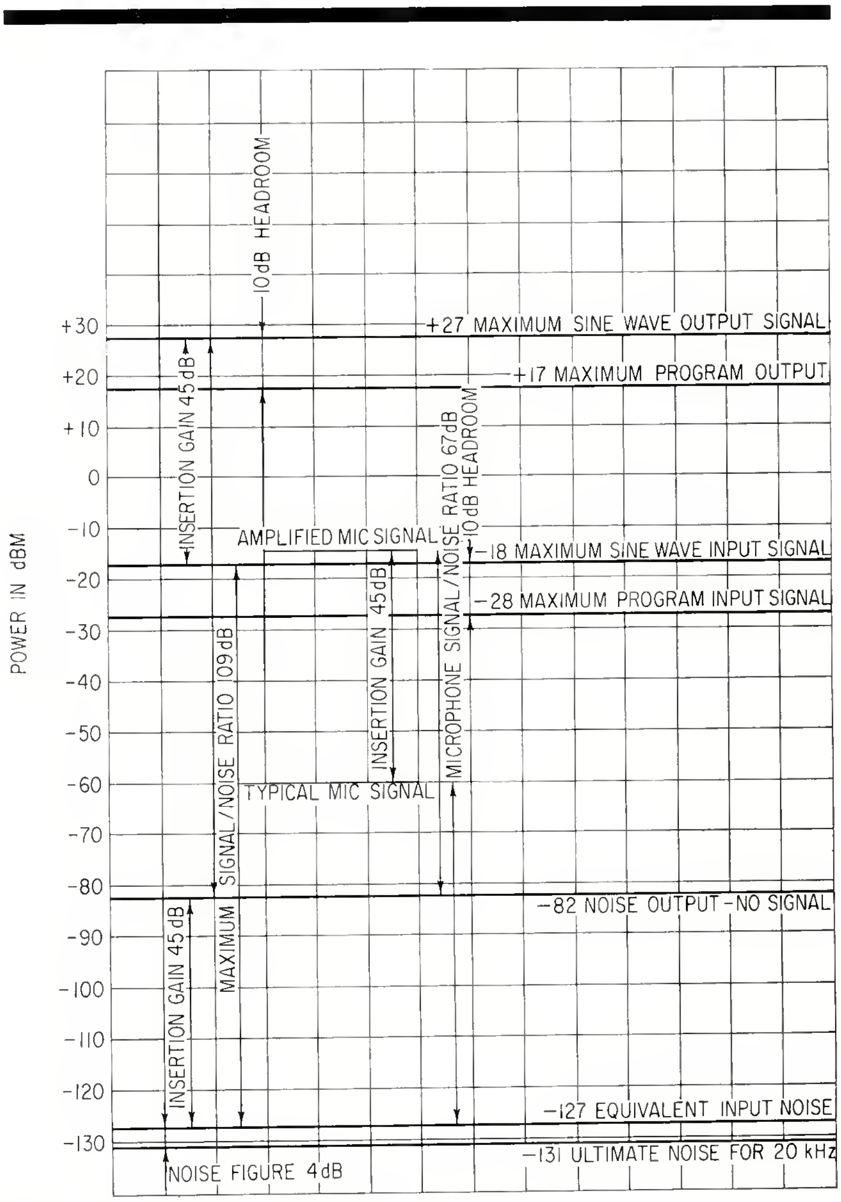

Figure 1. Typical signal and noise levels in a high-quality amplifier.

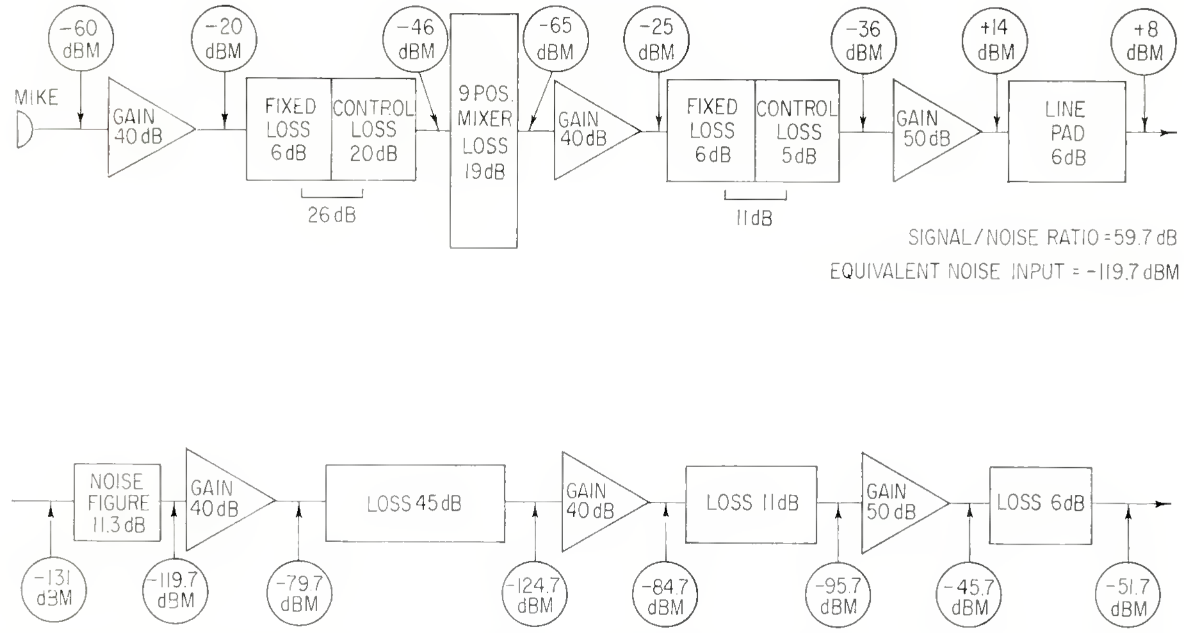

Figure 2. A simplified block diagram of a typical studio console intended to provide a line-level output. This is the signal path for one microphone. The lower portion of the figure shows the way ultimate noise proceeds through the circuit.

$$E^2 = 4kTBR \quad (3)$$

where B is the -3 dB bandwidth

If we perform a simple algebraic manipulation on equation 3, we arrive at the following form:

$$E_2/4R = kTB \quad (4)$$

This equation gives us the available noise *power* due to thermal noise. The available power means the power developed in a load resistor of R ohms fed from a generator whose open-circuit (no-load) voltage is E and whose internal resistance is also R ohms. This is the case in an "impedance"-matched circuit, where the load power from a given generator is maximum.

Since we now have an expression for available noise power in watts, we can convert simply to a perhaps more useful form.

It is customary, as suggested by Friis,[7] to calculate noise power at a temperature of 63°F or 17°C or 290°K. Suppose that the bandwidth is 1 Hz. Plugging these numbers into formula (4) we have:

$$\text{Noise power} = 1.38 \times 10^{-23} \times 2.9 \times 10^2 \times 1$$

or

$$\text{Noise power} = 4.0 \times 10^{-21} \text{ watts per Hz or } 4.0 \times 10^{-18} \text{ milliwatts per Hz} \quad (5)$$

Since this is an absolute power, we can convert to dBm:

$$\text{Noise power (dBm)} = 10 \log_{10} (4.0 \times 10^{-18})$$

$$\text{Noise power} = -180 + 6 = -174 \text{ dBm per Hz} \quad (6)$$

Equation 6 is quite useful since we can now quickly find the available noise power in dBm for *any* bandwidth by simply adding 10 $\log_{10}$ of the bandwidth in Hz.

For example, the available noise power for a 20 kHz bandwidth is:

$$\text{Noise power (20 kHz)} = -174 + 43 = -131 \text{ dBm} \quad (7)$$

The available noise power in a 15 kHz bandwidth is:

$$\text{Noise power (15 kHz)} = -174 + 41.7 = -132.3 \text{ dBm} \quad (8)$$

Equations 7 and 8 are extremely useful to an audio system designer since they give him the ultimate noise of his system and thereby determine the lowest value of signal which can appear in his system for a given signal/noise ratio, provided that the noise of the first amplifier is taken into account.

The ultimate noise is amplified by the first amplifier in the system as well as the desired signal (since the amplifier is unable to distinguish between the two). Since all electronic amplifiers introduce noise of their own, a figure of merit used mostly in r.f. work (which also is concerned with thermal noise) is the noise figure. This is the ratio of the actual noise output of an amplifier to the theoretical minimum. For example, suppose that we have an amplifier whose transducer gain is 50 dB and whose bandwidth is 20 kHz. The ultimate noise is -131 dBm for this bandwidth so that if we amplify it by 50 dB in a *perfect* amplifier (one which introduces no noise of its own) the noise output would be -81 dBm. This amplifier, however, (like all things from the hand of man) is not perfect and does introduce some noise, so that the noise output with no impressed signal measures -75 dBm. Thus the noise figure would be 6 dB since its output is 6 dB more noisy than a perfect amplifier.

Audio men have developed another method of measuring the merit of systems and amplifiers which is called equivalent-input noise. In the preceding example, the equivalent input noise would be -125 dBm, for if the amplifier were perfect, it would amplify noise of -125 dBm by 50 dB for a noise

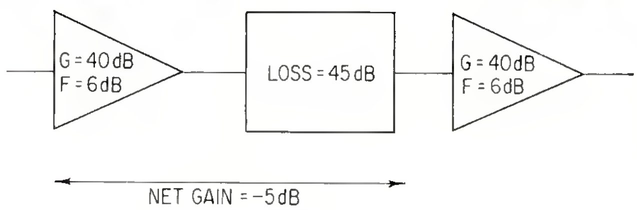

Figure 3. Here are the effects of the noise figure of the booster amplifier. Using equation (18) $F_s = 13.5$ and $F_{s(dB)} = 11.3$ dB.

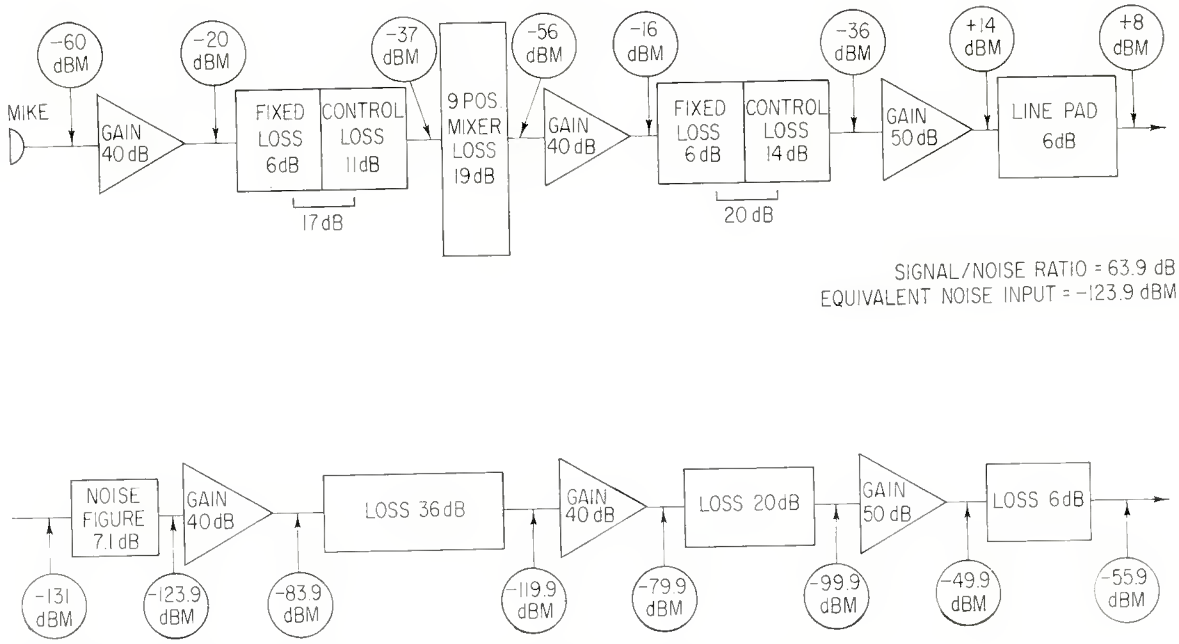

Figure 4. The same console as in Figure 2 but with rearranged control losses in the fader and master controls and with the same 68 dB gain. Again, the lower portion of the figure indicates the noise path.

output of -75 dBm. Thus the noise figure and the equivalent input noise really amount to the same thing.

It can readily be demonstrated[8] that the noise figure and gain of the first stage in a system is very important in establishing the over-all system signal/noise ratio. Suppose we have an amplifying stage whose gain is G, where G is a number rather than in dB. Actually G is

$$G = \text{antilog (gain in dB/10)} \qquad (9)$$

The noise figure has been defined- as:

$$F_1 = N_1/kTBG = \frac{1}{G} \times \frac{N_1}{kTB} \qquad (10)$$

where F is noise figure (Number)

N_1 is the noise output of the amplifier in watts

G_1 is the amplifier power gain (number)

Some of the power in N_1 is contributed by the amplifier so that this contribution is:

$$N_Z = N_1 - kTBG \qquad (11)$$

Substituting the value of N_1 as obtained from equation 10 in equation 11 and doing some factoring we have:

$$N_Z = (F_1 - 1)kTBG_1 \qquad (12)$$

Thus N_1 consists of two components:

$$N_1 = kTBG_1 + (F_1 - 1)kTBG_1 \qquad (13)$$

If we pass N_1 into a second amplifier whose noise figure is F_2 and whose gain is G_2, the output of the second amplifier will consist of three components:

$$N_2 = kTBG_1G_2 + (F_1 - 1)kTBG_1G_2 - \text{(contribution of amp 2)} \qquad (14)$$

The third component can be shown to be:

$$N_{a2} = (F_2 - 1)kTBG_2 \qquad (15)$$

The over-all system noise figure will be:

$$F_s - \text{Total noise output}/kTBG_1G_2 \qquad (16)$$

or

$$\frac{F_s - kTBG_1G_2 - (F_1 - 1)\,kTBG_1G_2 - (F_2 - 1)\,kTBG_2}{kTBG_1G_2} \qquad (17)$$

This equation simplifies to:

$$F_s = \frac{F_1 + (F_2 - 1)}{G_1} \qquad (18)$$

It may readily be seen from equation 18 that the system noise figure depends greatly upon F_1 but that F_2 not only has one subtracted from it but also it is divided by the gain of the first amplifier. Thus the noise contribution of the first amplifying stage usually establishes the system noise performance.

In the above example we have used the noise figure as a power ratio; we have also used the noise figure expressed in dB. The relationship is simple:

$$10 \log F_{\text{power ratio}}$$

Having had a dose of theory and mathematics to establish some background, let us now turn to some practical applications. Suppose we are called upon to design a system with a required signal/noise ratio of 65 dB. We have available an amplifier for the microphone signal whose manufacturer rates it as having an equivalent input noise of -125 dBm or a noise figure of 6 dB for a 20 kHz bandwidth. In order to obtain the 65 dB s/n ratio, the microphone signal must be 65 dB greater than the equivalent input noise or must be at least -60 dBm. Furthermore, in order to preserve this s/n ratio throughout the system, at *no* time must the signal level be allowed to drop below its previously amplified value because of such things as mixing network loss, insertion loss of ladder pads, excessive control loss, etc. Suppose also that instead of the first microphone amplifier (6 dB noise figure) we decide to go the cheapie route and we have another amplifier whose noise figure is 12 dB for an equivalent noise input of -119 dBm. Now we have a new ball game, and in order to establish a 65 dB s/n ratio the microphone signal must be no lower than -54 dBm. If our mic level is -60 dBm, then we can *never* be better than 6 dB out of specifications or said another way the best s/n ratio we can *ever* get is 59 dB.

The considerations for signal/noise ratio and some of the trade-off possibilities may be easily visualized from an examination of FIGURE 1. This figure represents the conditions for an excellent combination microphone-line amplifier which is rated by its manufacturer as follows:

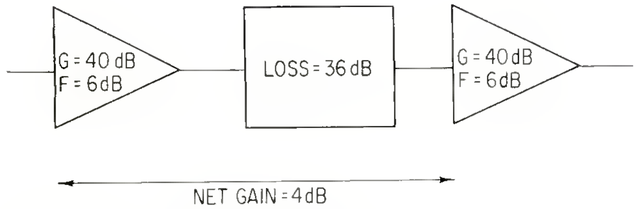

Figure 5. The calculations for the system noise in Figure 4. Using equation (18) we now find a F_s of 5.2 which translates to $F_{s(dB)}$ = 7.1 dB.

Transducer Gain	45 DB
Equivalent Input Noise	−127 DBM
Maximum output signal (sine wave)	+ 27 DBM

With this information, FIGURE 1 was constructed. Note that for a 20 kHz bandwidth the ultimate noise is −131 dBm and since the equivalent noise input is −127 dBm, the amplifier's noise figure is 4 dB. Since the noise of the equivalent input is amplified by 45 dB, the amplifier's noise output with no input signal will be −82 dBm. With the maximum sine wave output of 27 dBm, the maximum input signal level will be −18 dBm. Thus, with an input signal of this level, there is a potential signal/noise ratio of 109 dB as shown. If this amplifier is used as a microphone pre-mixing amplifier, and we use the generally accepted microphone level of −60 dBm, then the output microphone signal will be −15 dBm and the signal/noise ratio will be 67 dB as is shown. It can also be seen that it is possible to obtain a higher signal/noise ratio by using a higher input signal level. For example, a microphone whose output was −50 dBm would improve the signal/noise ratio by 10 dB to 77 dB. The euphoria of such signal/noise ratios is, however, quickly dispelled by the realization that the price paid is a reduction in upward dynamic range. For instance, the −60 dBm "average" microphone level gives an upward dynamic range of 42 dB of which at least 10 dB should be reserved for headroom thus restricting the available upward dynamic range to 32 dB. It can thus be seen that extended dynamic range is always at the price of signal/noise ratio and vice versa. These are some of the trade-offs that a system designer should consider.

Some of the principles considered in this paper and a practical application to a broadcast or recording studio can be effectively shown by a consideration of FIGURE 2. This is a simplified block diagram of a typical studio console intended to provide a line level output. The signal path for one microphone is shown, consisting of a pre-amplifier (gain 40 dB and noise figure of 6 dB), a ladder-type fader, a nine-position resistive mixer (only one input is shown), a booster amplifier (same characteristics as the pre-amplifier), a ladder-type master gain control, a program amplifier and a line isolation pad.

The total system gain is obtained by adding the gains of the three amplifiers; this is 130 dB. The system also has certain fixed losses; each ladder attenuator has a fixed insertion loss of 6 dB, the mixer network has a loss of 19 dB, and the line isolation pad has a loss of 6 dB, for a total fixed loss of 37 dB. Thus the available net gain is 130 − 37 = 93 dB.

Now a console is never operated "wide open"; there is always some control loss inserted in the fader and master attenuators. In FIGURE 2 the fader control loss is 20 dB and the master-control loss is 5 dB; thus the usable gain is 68 dB. If we assume a microphone signal level of −60 dBm, then the output will be +8 dBm. FIGURE 1 contains balloons which show the signal levels throughout the system.

At the bottom of FIGURE 2, we have shown the same system but this time we are interested in the way that the ultimate noise proceeds through the console. We have assumed a 20 kHz bandwidth so that the ultimate noise is −131 dBm at the microphone. The next block is labeled "Noise Figure" and this deserves comment. It will be recalled that the noise figure of the preamplifier and the booster amplifier were given as 6 dB each (a respectable value). It will be noted that the combined losses of the fader control and the mixer network add to 45 dB which is *greater* than the transducer gain of the pre-amplifier. Thus the signal emerging from the mixer network (top of FIGURE 2) is 5 dB *less* than mic level; and the noise (bottom diagram) is now back in the region of the ultimate noise. It is necessary, therefore to consider the effects of the noise figure of the booster amplifier. This is done as is shown in FIGURE 3. Here we apply the formula of equation (18) covering the system noise figure produced by two amplifiers in cascade. From the mathematics of FIGURE 3, it will be seen that the effects of the 45-dB loss in the fader and mixer are to increase the system noise figure to 11.3 db from the original 6 dB of the first amplifier. The balloons on the noise diagram give the noise levels throughout the console, with the output noise being −51.7 dBm and a signal/noise ratio of 59.7 dB. The equivalent input noise is −119.7 dBm.

If we turn to FIGURE 4, we have the same console but with this variation: we now have rearranged the control losses in the fader and master-gain controls, but maintaining the same over-all gain of 68 dB. FIGURE 5 shows the calculations for the system noise figure in this case. Since the fader and mixer losses are now 36 dB, we have an over-all net gain of 4 dB between the first and second amplifiers. The system noise figure calculates to 7.1 dB, a noise degradation of 1.1 dB over the original 6 dB. In this case, the signal/noise ratio is now 63.9 dB and the equivalent noise input is −123.9 dBm.

The moral is clear: In order to preserve, protect, and defend the original signal/noise ratio established by the difference between the input signal level and the equivalent input noise, it is desirable and necessary that the input fader be run quite "wide open" and the master used to establish the output level. In *no case* should circuit losses permit a signal to emerge at a lower level than at some earlier part of the system.

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Multi-Purpose Halls Present Problems

Here is one solution to the problem of installing versatile sound systems in a hall that was not designed with listening acoustics in mind. The system described serves well regardless of the use of the hall and the need for speech-only or wide-range music dispersion.

MULTI-PURPOSE HALLS which are becoming more and more frequently seen all over Europe, have very considerable advantages for local authorities or the investment companies who build them, for they can be used for trade fairs, exhibitions, ice hockey, figure skating, basketball, and also for large meetings, as well as music hall and theatre productions.

But the acoustics of such large buildings are hardly ever ideal. The great volume of such halls, usually about forty thousand cubic metres, normally tends to cause very considerable echo, with the sound reverberating away under the roof. It is extremely difficult to attain uniform distribution of direct sound in such large areas where architectural considerations have had to dominate and where it is impossible to cover such areas with understandable sound by conventional means because of the wide variety of use to which they will be put.

One of the most recent of these types of building to come to our attention is the one at Bergen in Norway. There was considerable discussion about the advisability of adopting the distributed sound system, or alternatives.

The distributed system uses many loudspeakers, suspended from the ceilings, or it is possible to have a central distribution system with the sound coming from one or two points. The former solution presented considerable difficulties because of the architectural nature of the hall and because such an installation, if adopted, would considerably limit the uses to which the hall could be put, because speech intelligibility over such a system is not ideal. As a result, the system adopted at Bergen uses central distribution of sound as the

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only way of providing the flexibility necessary for this multi-use hall.

In a hall which is square or circular a single distribution point is usually sufficient (and certainly desirable) because from each loudspeaker point in such a system there is a symmetrical angle of sound coverage. In a rectangular hall where there is a considerable difference between the long and short sides, it is difficult and sometimes impossible to adequately cover the hall from a central sound point, avoiding echoes arriving at the audience later than direct sound.

In the case of Bergen, therefore, two central distribution points with clusters of loudspeakers were chosen which would, separately and together, satisfy the requirement of ninety decibels sound volume uniformly over the whole area of the hall.

These two clusters of loudspeakers consist of a series of speaker columns, each of which covers a space within a ninety-degree angle over its area of the hall, and each column speaker is specially designed for this particular hall so that each provides adequate coverage for the tiers of seats and the gallery in its quadrant.

To deal with the ice rink or actual floor of the hall, ordinary loudspeaker systems in inverted-pyramid shapes are used. These provide a circular coverage and experience has shown that two such clusters operating together will provide a sound level in a hall of this type that is more or less hemispheric in shape, so that the sound covers the center area of the floor of the hall as completely as the outer edges of its range. The result is that there is uniform emission of sound to ensure that people will hear clearly.

Each column of speakers was designed from the point of view of the mechanical size necessary for the angle of distribution demanded and with regard to suitability of output to the field of coverage. The type of loudspeaker was chosen on the basis of those frequencies which demanded priority in these conditions.

The column speakers and the standard loudspeakers at the base of each cluster are used for speech only. To deliver first-class reproduction of music there is linked to each cluster a large bass cabinet high up under the roof with wide dispersion angles to ensure the extensive distribution of the deep notes and afford complete bass coverage of the whole hall when music is relayed through them.

These bass loudspeakers can be cut off separately, and they cannot be switched on at all while microphones are in use. Individual column speakers or clusters can also be cut out so that the hall is acoustically suited to the various purposes for which it will be used.

Figure 1. The control panel at Bergenhalls overlooking the main arena.

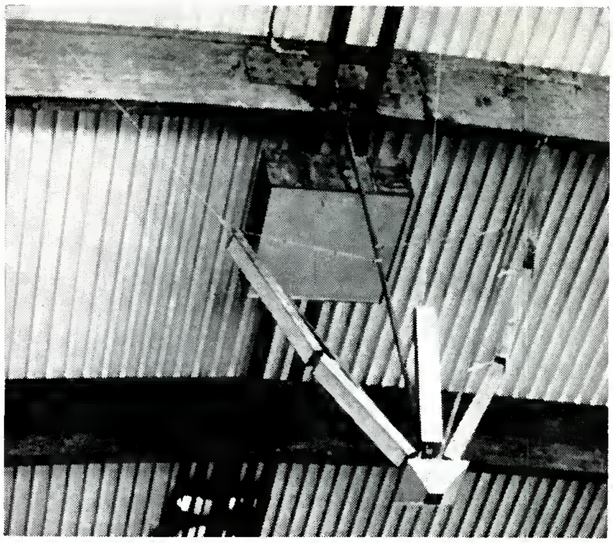

Figure 2. One of the loudspeaker columns in the ceiling. Note the bass speaker close against the roof beams.

During ice-hockey matches, for example, there will be no need to distribute sound across the ice, and if there are only a few spectators the column speakers which distribute the sound to the upper sections of the stands will not be necessary and they will, therefore, be cut off. In some instances only one cluster may be in use at any one time.

In addition to the loudspeaker system in the hall itself there are loudspeakers in the changing rooms, cafeteria, and the restaurant. A large mixing desk with amplifier racks is so situated that the operator at the mixing desk has a full view of the hall at all times so he can make necessary adjustments quite easily. (Figure 1) The desk carries six integral amplifiers one of which acts as a preamplifier and mixer. There is an amplifier for each cluster, an amplifier for the bass loudspeakers and two amplifiers for cloakrooms, cafeteria, restaurant and outside loudspeakers.

Also in the desk is a radio tuner, a record player and a tape deck, as well as a microphone. By means of the control loudspeaker and vu meter both program sources can be checked before the program is released into the hall. In addition level and quality can be monitored during transmission at any time.

When it is necessary to make announcements from the management office, or information kiosk, or from the commentator's position, all outgoing music is automatically faded down during the announcement and then brought slowly back up (automatically) after the announcement is concluded.

Because the architect and the acoustics consultant worked together carefully, the ceiling of this hall affords better sound attenuation than is usual in such large buildings. While the installation was undergoing final tests and adjustments it was therefore found that the echo interval and reflection were already down to a minimum. Because calculations of the loudspeaker installation were based on the attenuation factor of the ceiling it has been found that the installation satisfies the requirements stipulated for in the beginning, namely a very uniform distribution of early sound, practically no reverberation or echo.

Even when the microphone is in use in the middle of the hall or in its most difficult areas, quite adequate sound volume can be achieved without acoustic feedback. Speech comes over clearly and yet there has been no concession to the high expectations for the reproduction of music.

Implications of the Low-Noise Background

EDWARD TATNALL CANBY

It is taken for granted by audio engineers that noise levels should be reduced to their theoretical levels. But what happens when you reduce the noise content of a record below the point at which ambient noise exists in the concert hall. The author examines some of the philosophical aspects of the new techniques of noise reduction.

THE TOTALITY OF THE AUDIO REPRODUCTION CHAIN has gone through three major phases of special concern in the burgeoning twenty-year period since that unhappy phrase "hi fi" became a catchword for public interest in better audio. As I see it, right now we are deep in the third of these phases, one that in a curious way is not directly concerned at all with the audio signal.

It takes, perhaps, an outsider like myself, who is also by sheer contact a sort of audio insider, to formulate these concerns; for they are not reflected in statistics nor in accounts of technical progress in engineering language. Indeed, for the audio man they may be hard to recognize at all as he looks backward over the maze of interlocking developments during these many years. But the consumer's eye, I think, sees the picture clearly, if only for being outside of the forest, away from the technicalities.

Let me look at these three in general terms. *First,* soon after the war and as hi fi began to formulate itself in the public gaze (*Life Magazine* made it official—was it 1948?), our primary concern was the expansion of the audio bandwidth, not in the lab but in the outgoing product. Wide-range sound was the popular terminology. Not that other phases of audio were ignored. Far from it. Yet nevertheless all of us, engineers, publicity, and consumers, were then first-of-all conscious of this important aspect of progress. For, as we remember, practical bandwidths outside the lab had been very severely restricted up to that time, both in the high end, generally fading to nothing by 4 kHz, and in the bass, where in spite of the solid boom of the electric juke box the actual lows were not very low. Extended-range sound in practical form was an exciting development in the late forties and very early fifties even with the rude penalties we had to pay in terms of hiss and distortion. We paid them gladly—for awhile.

But about the time that those once-dramatic figures "15 to 15,000 cycles" gave way to a more prosaic "20 to 20,000" for almost any equipment you could advertise, professional or consumer, there came a new emphasis of a soberer sort. It was, of course, no longer on tonal range, which had become a commonplace, but on distortion (or rather the reduction of distortion in all its forms). You may supply your own milestones for that long-continuing major concern, this *Phase Two,* which subdivided itself more-or-less chronologically according to the simpler and more complex distortion forms. (First, simple harmonic distortion, then later there was widespread preoccupation with intermodulation and finally, along with stereo sound, a still more subtle concern over such matters as exemplified by phase distortion.)

Many a product and popular phrase comes to mind as we recall this major phase of audio progress in the consumer and professional areas, as one form of distorted signal after another was brought to heel in both electrical and mechanical terms. At random, I think of the great triode excitement and the Williamson amplifier, the hot stylus breakthrough, the Leak "Point 1", proclaiming in its very name the triumph over distortion; there were the new and slightly more meaningful catch phrases, the wide use of "flat", or "within X dB" (put those two together and you have something), replacing earlier claims of tonal range alone. (Let us put aside questions of meaningfulness in these popular "specs". I mention them as symptoms of the major concern of the time.)

Phase Two was immensely strengthened, as well as lengthened, by the solid-state revolution which arrived just in time to bring further and more dramatic improvements in over-all distortion at every audio quality-level, just when tube cir-

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cuitry had begun to reach down into the esoteric in these respects. (Tube developments still go on, but they are no longer a major area of concern.) Solid-state audio carried our special interest in ever-lowered distortion to an extreme, and then on to its ultimate decline. Today, there really isn't much distortion left to talk about, at least in the purely electronic areas of audio.

And so to the *third* and present concern, coming after these two. With an audio bandwidth available, at least rhetorically, from zero to infinity in electric terms, with electrical distortion down to milli-fractions of almost nothing, what is left?

There is, of course, the electro-mechanical sound transduction, principally in the loudspeaker. (Microphones, phono cartridges, and cutting heads are relatively well ahead.) But though speakers have been steadily improving in recent years towards that *gray-flannel-suit* uniformity of non-coloration that I once suggested as a highly desirable end, there is (if I may dangerously guess) no major large-scale breakthrough now at hand. In loudspeakers, as in TV tubes (until the flat wall tube arrives) we inch forward confidently but we do not break through. Ours is not yet a speaker era in terms of overriding concern.

What is left for *Phase Three*, then, is quite literally residual noise. Noise over and beyond the signal, noise unintended, uninvited and non-signal, the negative aspect of audio technology. That is where we are now battling, and breaking through. In a sense we have put aside the audio signal itself; it's doing OK, thank you. Our major interest has become the signal's framework, and things have been that way for a number of years already.

Someone recently remarked to me that if the Dolby S/N Stretcher system had not appeared several years ago, our present interest in the fantastically low-level sound residues would not have been aroused; we would have gone along as before, accepting things as they were, and quite satisfied with the status quo. Not true! I prefer to think that Ray Dolby is one of those engineers who hit the right nail on the head at precisely the proper moment.

True, the Dolby system has brought this concern with infinitesimal (but highly audible) low-level noise to dramatic public notice, and to the manufacturers of records and purveyors of tapes (not to mention the operators of noisy communications circuits) within the larger audio field. Dolby has promoted a one-man revolution whereby Dolby-ized tapes are already near-standard in much audio recording, to the tune of incredible outlays for Dolby-built equipment. But the low-noise field is far from a Dolby exclusive.

We must remember that in all such phases of widespread technical concern there is a convergence of many lines of thought and development, seemingly out of hundreds of unrelated enterprises. Some years before Dolby, 3M's low-noise Dynarange 200 series recording tape brought the matter to professional interest. It was about time — for the advances in other audio areas were bringing the very low-noise levels increasingly to the fore; they were beginning to bother us.

Typically, the 3M product was part of a side operation, the development of the Revere automatic tape cartridge player using a narrower slow-speed tape. But it was in the standard tape size that the 3M coating made its significant impact. Suddenly we discovered that, after all, there was still room for dramatic s/n improvement even within the strictly competing parameters that must be juggled in tape formulations. Might there be other advances of the sort elsewhere? We hoped so. Meanwhile in Europe Philips took over the narrower tape for its successful cassette, where low noise is vitally important. Wheels within wheels. Now we can anticipate Du Pont's chrome tape for a further move in expanding the low-noise potential; here also the advance was a byproduct of a wholly unrelated development at Du Pont. And so it goes.

So we have all these low-noise innovations, quite aside from Dolby, and any seasoned audio engineer can add a dozen to the list. Low noise is, so to speak, in the air. It is crucial in every phase of current research as information densities increase, as the physical media with which we work get smaller, go slower and move nearer to the noise danger threshold.

It occurs to me, perhaps anticlimactically, that low noise has even made macro-mechanical breakthroughs. The two most recent turntables that I have acquired are so silent that, time after time, I leave them running for hours by mistake. Technically a low-noise disadvantage, if a minor one. One table is built into a popular "library" hi-fi system which in itself is electronically so quiet (and has such feeble low-heat pilot lights) that I find this sort of accident a real inconvenience. That machine has uselessly churned thousands of circular miles with all its electronics on and ready to go and never a trace of audio, for days on end.

Laying aside these unimportant problems, and postponing for the moment some more serious questions in music listening, consider the usefulness of the new low-noise or no-noise background. I scarcely need to recapitulate them. There is the increased dynamic range on tape first of all, anywhere from 3 to 15 dB more before the mud level is reached, and the greater clarity of a non-muddied signal when the thin veil of almost inaudible background noise is reduced. There are the special values of Dolby's crucial reduction of print through after the fact, one of the major reasons for the system's success among recording companies. Also the much improved copying accuracy in an area where noise build-up has long since replaced distortion as the worst factor in deteriorization from copy to copy. There is the practicability of Dolby-izing for low noise within the home-type tape recorder, as now pioneered by KLH, and the future possibility for low-noise disc records, the disc itself included within an extended Dolby stretch. These and other valuable results from lowered noise are understood by every audio engineer. And because low-noise improvement in one area exposes residual faults in other areas, the competition now extends into every part of audio and off into such important side-areas as plastics, where the disc makers are sweating out the new demands put upon their pressing plants. This is surely a wave of major concern, this *Phase Three*, and the only strong arguments I have heard about it among engineers concern procedures and principles: are such ingenious devices as the Dolby merely shortcuts?

My own feeling on that score is that, just as the road to Hell is paved with good intentions, so the way to engineering perfection is lined with highly successful compromises. One could mention in audio the non-linear disc record and its wrong-tracking arm (the cylinder does better on both counts), still virtually standard after 75 years; the trickly preemphasis curves for discs and tapes and for FM broadcast; even — if we must argue — the extensive use of feedback around various segments of amplifier circuitry. All of these are to a degree crutches, practical means towards approximate perfection, done with the least possible complexity and the most reliability, useful on the largest scale and with economy of cost and trouble. On a high level, to be sure, the Dolby system is indeed a relative of these, and not too distantly related to the preemphasis-deemphasis principle.

And yet one can sympathize with the purist argument of David Hancock, independent New York recording engineer, who feels that low-noise improvements should be made without artificial aids, in the really basic areas where the problems are actually created. Maybe impractical, but his point of view is a good way for a reputable engineer to feel. Those who agree with him will continue to safeguard audio standards from over-gimmickry.

Having argued these points, we find ourselves with the plain fact of dramatically improved s/n ratios today, in our newest state-of-the-art tapes and in the best of the records being made from them (with concomitant disc improvements to match). Whatever the technical means, the technical values are already enormous. And so I must turn to what is the only further consideration of real importance — the aesthetic. It is a curious convention among audio engineers that content is not their concern; their business is wave shape. One might suggest that the wave shape *is* the content, for there is no use prying them apart. They exist as one.

Moreover in terms of audio *consumption* the content of the audio signal is all important, not to say omnipotent. There is no audio which is not consumed, one way or another, by some waiting ear tuned to the given message. To put it baldly, the message is the medium. The impact of any audio signal depends on what sort of message it conveys; and each message has its own demands.

And so I must prepare to take a flyer into the aesthetics of low-noise audio, particularly in my own province, home-reproduced music, an area of major importance in the audio field. In music, low noise is not the uncomplicated boon you may think, though its advantages are bound to be great. Do we, for instance, really want that wider available dynamic range that is the first direct result of a more silent low end? Most engineers will take it for granted that we do. Not always, I say. A good deal of music listening is merely confused, in more ways than one, by a larger dynamic swing from loud to soft and an inaudibly low level of background noise.

Before launching in detail into this curious aspect of low-level noise improvement, I must conclude by calling attention to the human ear's astonishing abilities at ultra-low sound levels. If it were not for that, the present furor over microscopic sound residues could not exist. But the fact is that the further we lower the general background sound level, the more pronounced is the unmasking of incredibly tiny residual unwanted noises and the more acutely do these impinge on our conscious hearing. The loud scratch of the old 78 acoustic shellac was taken in stride by a million ears, but the faint traces of undesired non-signal that now remain to us are painfully audible out of all proportion to their intensity. That is the rub!

But as I hope to indicate in a future installment, there is a paradox here that I suspect has so far gone unnoticed. Often these tiny residual sounds serve as important, if unconscious, aesthetic indicators to the home-listening musical ear. If we reduce them to actual inaudibility we may do so at no small risk to the aesthetic message in the recording and to the listener's musical satisfaction.

Possibly the coming era of no-noise (i.e. no-background) sound reproduction will mean the retraining of a whole generation of ears, to accept new clues to musical meaning via the loudspeaker. It is that serious. And with these we may see an extensive revaluing of the kinds of music best suited to home listening that could alter the entire recording scene. All from the elimination of a few infinitesimal sound-residues! There is more to this *Phase Three* than may be apparent.

Picture Gallery— NAB Convention

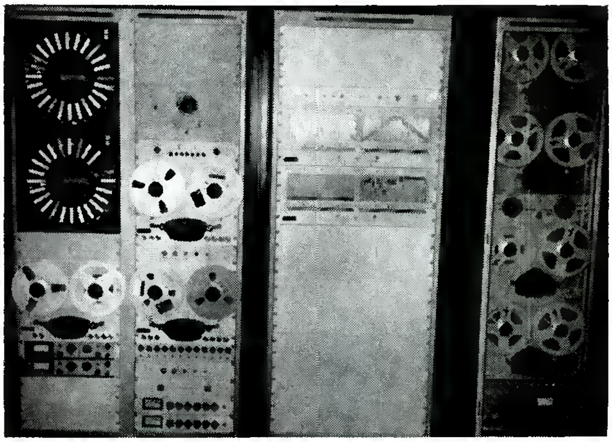
When there are almost a hundred and a half exhibitors spread over two Washington, D.C. hotels, the Sheraton-Park and the Shoreham, and four days it's a busy time indeed. This year's convention and exhibition was held from March 23rd through the 26th. The time was hardly enough for us to see all we wanted to, much less push our cameras everywhere they ought to have gone.

What follows then is a sampling of the equipment on display. Each of the photographs has a Reader Service Number which can be circled on the post card bound into the back of this issue to secure detailed information directly from the appropriate manufacturer.

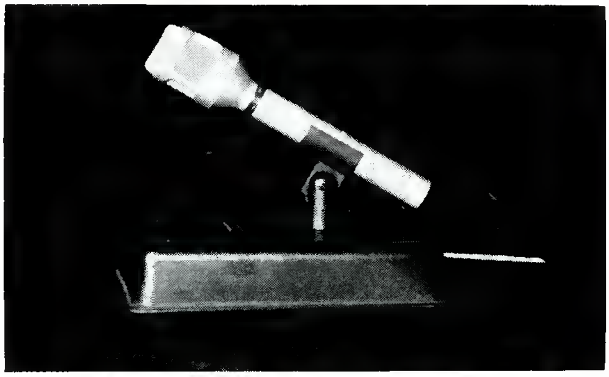
If any trend can be described for the show, it is that the movement toward automation of broadcasting stations continues. Not only complete station setups, but sub-systems are appearing with remote control or self-operation features. Nevertheless, there is a continuing flow of more sophisticated but conventional equipment — all solid state — to cover every possible application.

It should be understood, of course, that our pictures only show a sampling of what each manufacturer has. It can be assumed that if one console or microphone is shown, there are others to cover a variety of applications.

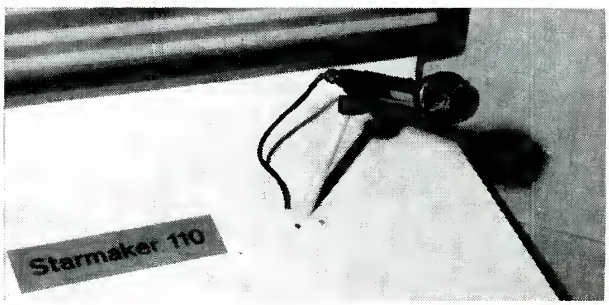
Vega Pro 55 FM wireless microphone system. Circle 15 on Reader Service Card.



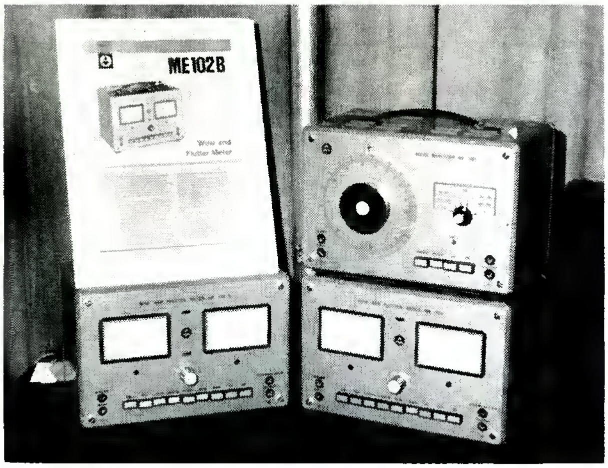
In ample evidence this year again — completely automated broadcast stations. This example used **Tape-Athon** and **Carousel** cartirdge tape machines. Tape-Athon information circle 80 on Reader Service Card.



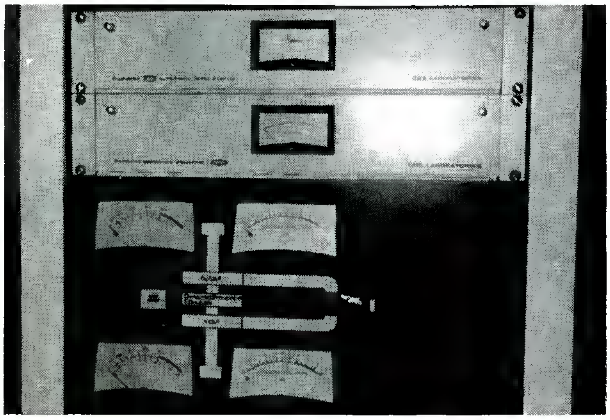
Shure SM53 dynamic unidyn microphone. Circle 13 on Reader Service Card.



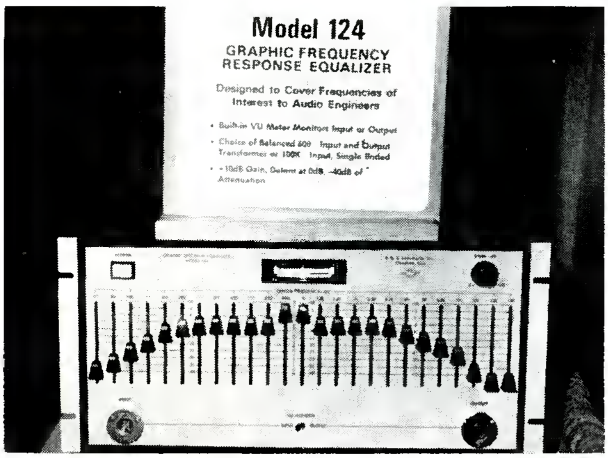
RCA Starmaker 110 dynamic cardioid mic for close work. Circle 18 on Reader Service Card.



Gotham Audio ME102B, ME104 wow/flutter meters and **ME301** wave analyzer. Circle 29 on Reader Service Card.



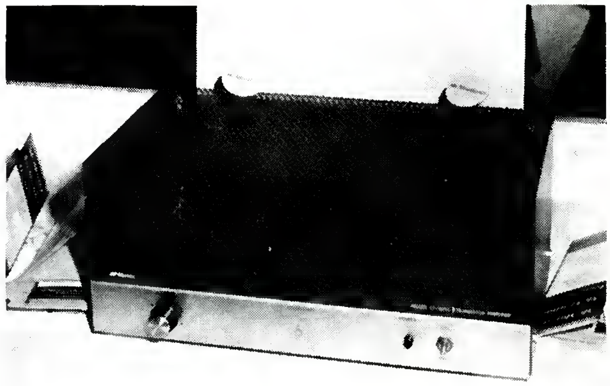
CBS Laboratories Audimax 111 and dynamic presense equalizer. Circle 34 on Reader Service Card.



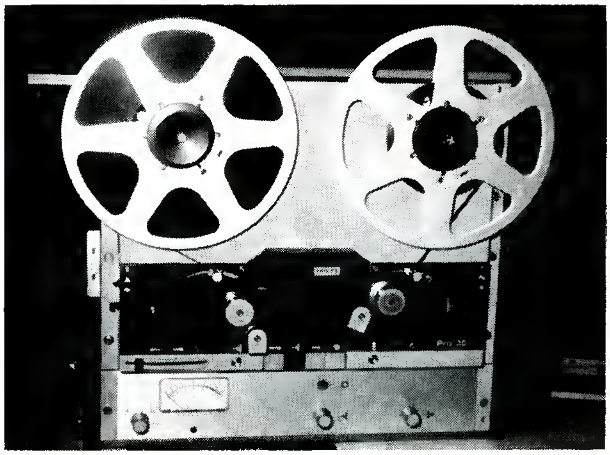
B & K Model 124 graphic equalizer. Circle 20 on Reader Service Card.

QRK Custom 12 two-speed turntable. Circle 10 on Reader Service Card.

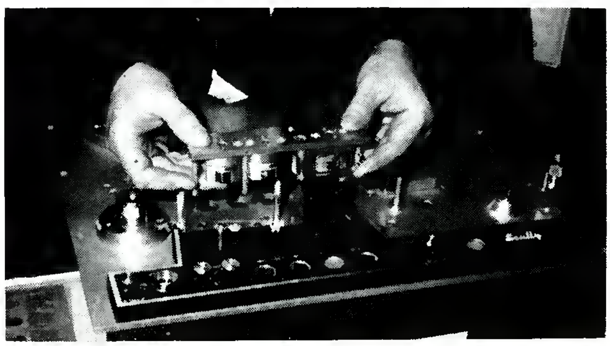
Gray ST-33 turn table with **model 303 arm.** Circle 24 on Reader Service Card.



McMartin LT-500A solid-state amplifier. Circle 16 on Reader Service Card.

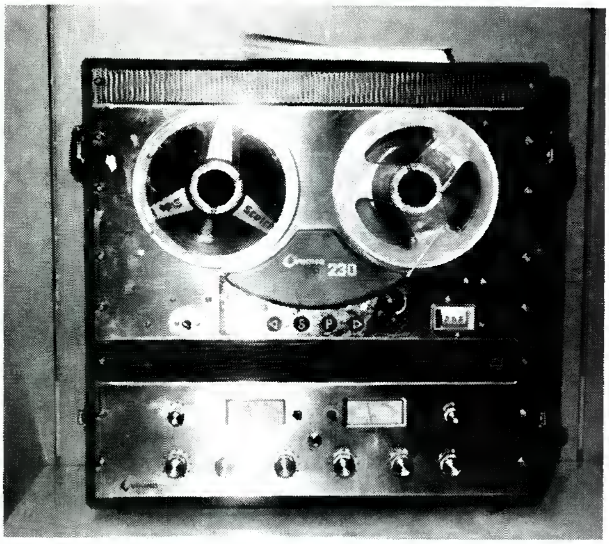


Philips PRO-36 3-speed NAB-reel recorder. Circle 17 on Reader Service Card.

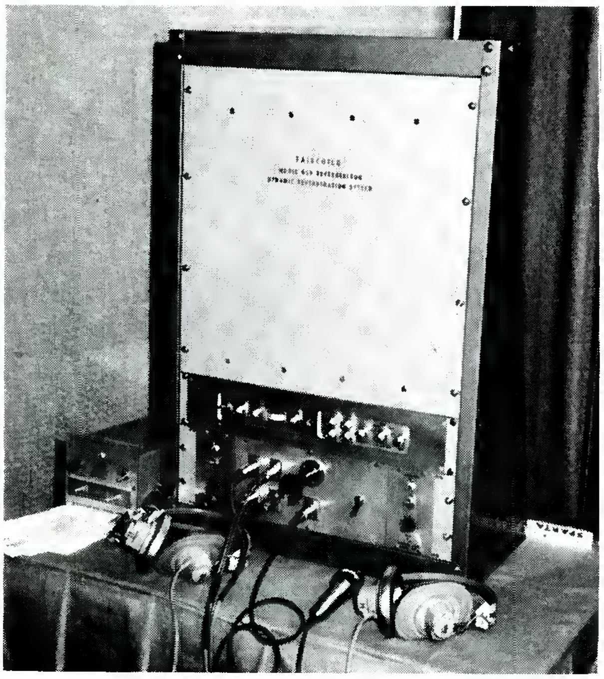


Nortronics replacement head assemblies for pro recorders. Circle 26 on Reader Service Card.

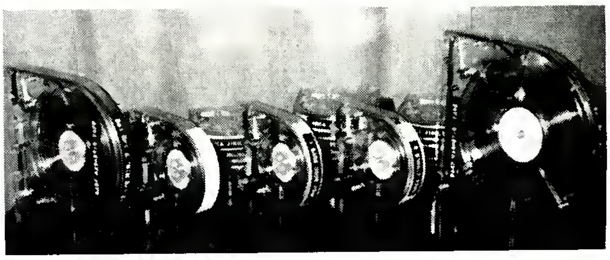
Ampex 440 series 4-track, ½-inch tape recorder. Circle 14 on Reader Service Card.



Viking model 230 stereo recorder. Circle 23 on Reader Service Card.



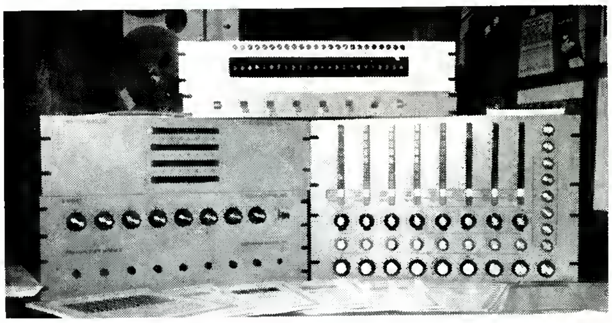
Fairchild 659 Reverbertron reverb system. Circle 11 on Reader Service card.



Marathon differential-pulley continuous-loop cartridges. Circle 32 on Reader Service Card.

Belar one μsec adjustable peak limiter. Circle 25 on Reader Service Card.

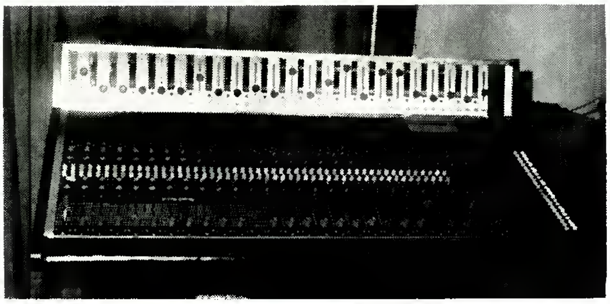
SIS ten-mixer position broadcast console. Circle 19 on Reader Service Card.



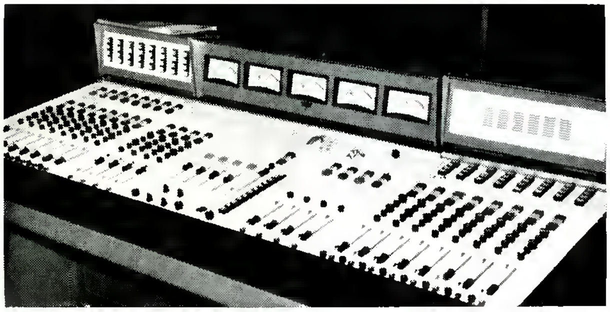
Suburban Sound equipment for console installation. Circle 22 on Reader Service Card.

Gates Radio 10-channel broadcast console. Circle 31 on Reader Service Card.

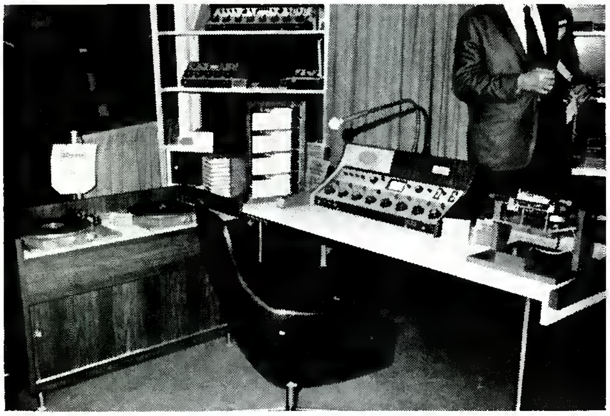
Ward 12-channel small console. Circle 27 on Reader Service Card



Altec-Lansing recording console with discrete lites for VU indication. Circle 30 on Reader Service Card.

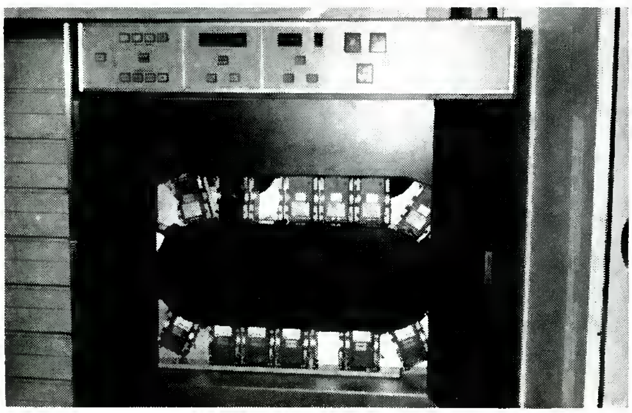


McCurdy Moduline record or broadcast consoles, up to 30 channels. Circle 33 on Reader Service Card.



Sparta 8-channel console with other distributed equipment. Circle 28 on Reader Service Card.

Collins Twintape record/play/dub cartridge machine. Circle 12 on Reader Service Card.



One of the most crowded exhibits at the show. A video tape cartridge changer from **RCA.** Circle 81 on Reader Service Card.

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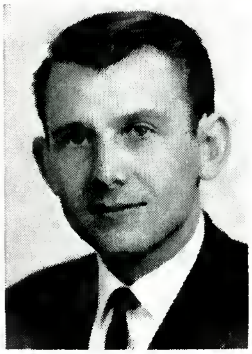
Ekornhoel

●**Tandberg of America,** independent distributor for the U.S. of Tandberg products has been purchased outright by **Tandberg Radiofabrikk** of Oslo, Norway. **Mr. Kjell Ekornhoel,** an engineer with the parent firm in Norway, has been appointed president and chief executive of the American organization. The founder and former president, **Eric Darmstaedter,** will remain associated with the firm as a consultant and adviser on a life-long basis. According to a company spokesman, **Robert J. Bowman, Jr.** has been appointed vice president and sales manager for the firm, while **William Hausman** will be director of educational sales. In announcing these new appointments, Mr. Ekornhoel said that the company hopes to establish a close rapport with the American market and intends to introduce progressive policies resulting in better service and products. Tandberg of America, Inc. distributes the Tandberg tape recorders, language laboratories, and speakers, as well as SolvSuper radios and other audio products.

●An announcement from the **NAB** indicates the formation of a liason committee to provide a channel of communications between that organization and the **Record Industry Association of America** (**RIAA**). The joint committee will concern itself with those matters involving either or both industries where one might assist the other. In this manner a line of communication will be established so that each interest knows what the other is thinking and doing, thus enabling a more effective area of mutual planning to meet the challenges of both programming and production of recorded music as used in radio.

●**Micom, Inc.** a manufacturer of precision electronic instruments has changed its name to **Data Measurements Corporation,** according to an announcement from **John S. Ames,** president. According to Mr. Ames, the name change was made because the original company name does not reflect the activities of the company. The firm's instruments are used to measure, calibrate, and analyze electronic signals produced by other instruments.

●We take note of the announcement that **Philips Gloeilampenfabricken** of Eindhoven, Holland has been licensed by **DuPont** to produce chromium dioxide tape in Europe. No terms nor the timing of production were disclosed.

HOWARD M. TREMAINE

Howard Marsh Tremaine died on May 20th. At the Hollywood AES Convention held just a few days before, he had held forth at the Howard W. Sams books booth which was directly next to our db Magazine booth. In those few days, I came to know and admire him greatly. That he was a man of great knowledge and experience is known by everyone. It was my privilege to discover him to be a man of deep humility and charm. We spent a good deal of time talking — or more rather, with him talking and me listening — for he had much of interest to say. He was 68 at the time of his death, but his mind was one of inquisitive youth. I know that there are many that share with me a terrible sense of loss at his passing.

Here are some details of his fruitful life so kindly supplied to us by his wife:

As a child he travelled with his parents in show business under the name *The Musical Tremaines.* At that time he acquired a book written by Marconi on wireless that was to shape the rest of his life. Along with a friend he became a wireless operator with the call letters 3BQ; this was prior to World War I. After the war he moved to Chicago, where he used his show-biz experiences to organize an all-saxaphone band that would go up on the roof of the building where their music was picked up by telephone lines and relayed to the local radio station for broadcast. In 1923 he moved to Hollywood. There he joined the Jesse Lasky Famous Players as a recording and development engineer. He was later associated with the Victor Talking Machine Co., when they developed their first electrical reproducing machine. In collaboration with J.N.A. Hawkings he developed the recording and reproducing equipment for the classic Walt Disney production *Fantasia.*

From 1941 to 1946 he served in the U.S. Navy as an electronics officer, with the rank of Lt. Commander. From 1946 to 1951 he operated a school of audio engineering, which closed when he suffered a heart attack.

In more recent years he had been chief of the sound division, Lookout Mountain Air Force Station, Hollywood. In 1962 he moved to West Vancouver, B.C. Canada where he supervised the construction of the Panorama Film Studios, and remained on as their consulting engineer.

Mr. Tremaine was a past chairman of the L.A. Section of the AES, and West-Coast vice-president and treasurer in 1952. He was made a Fellow of the AES in 1955. He was active in other organizations as well.

Howard M. Tremaine will best be remembered for his writing. He had authored eighteen papers and five books. At his death he was proofreading the galley pages of his sixth book, the second edition of the *Audio Cyclopedia* to be published by Sams later this summer.

L.Z.

Acoustic Research AR-3a speaker systems are important professional tools to composer/arranger Don Ellis.

Don Ellis creates music that ranges from the ancient sitar to a novel four-valve quarter-tone trumpet specially made for him. His work is well exemplified by **Electric Bath** (Columbia 9585), which was Album of the Year (1968) in **Down Beat,** placed second in **Playboy's** annual poll, and third in **Melody Maker;** the record was also nominated for a Grammy Award.

Mr. Ellis' high-fidelity system in his studio consists of an AR turntable, a Bogen-Lenco B62 turntable, an AR amplifier, a JBL 600 amplifier, a Koss Pro 600A headset, Revox and Crown tape recorders, and a pair of AR-3a speaker systems.

Mr. Ellis advises AR that the turntables, amplifiers, and tape recorders are all capable of highest-quality reproduction, so that making comparisons of different tapes and records can be done dependably with any of them. However, he finds that only AR-3a speaker systems are accurate enough to use in his work.

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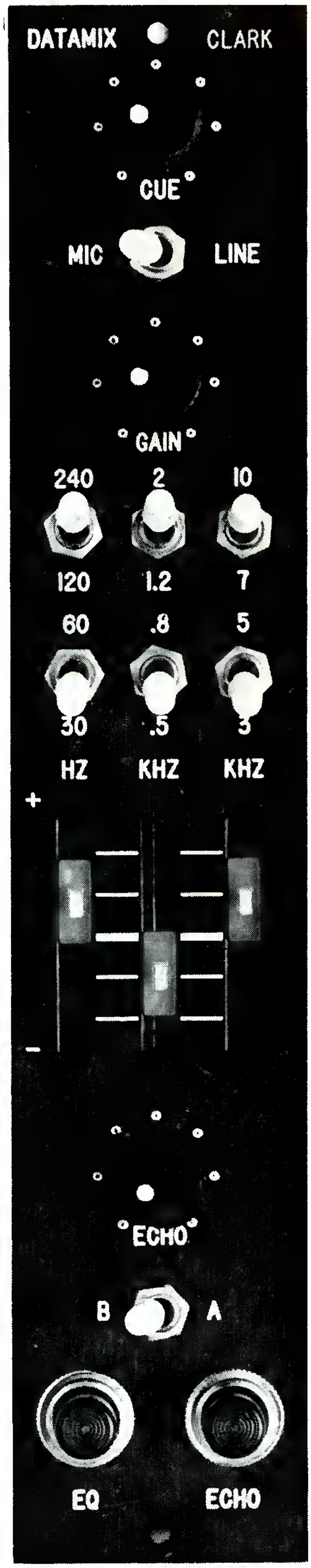

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