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THE SOUND ENGINEERING MAGAZINE

MARCH 1968

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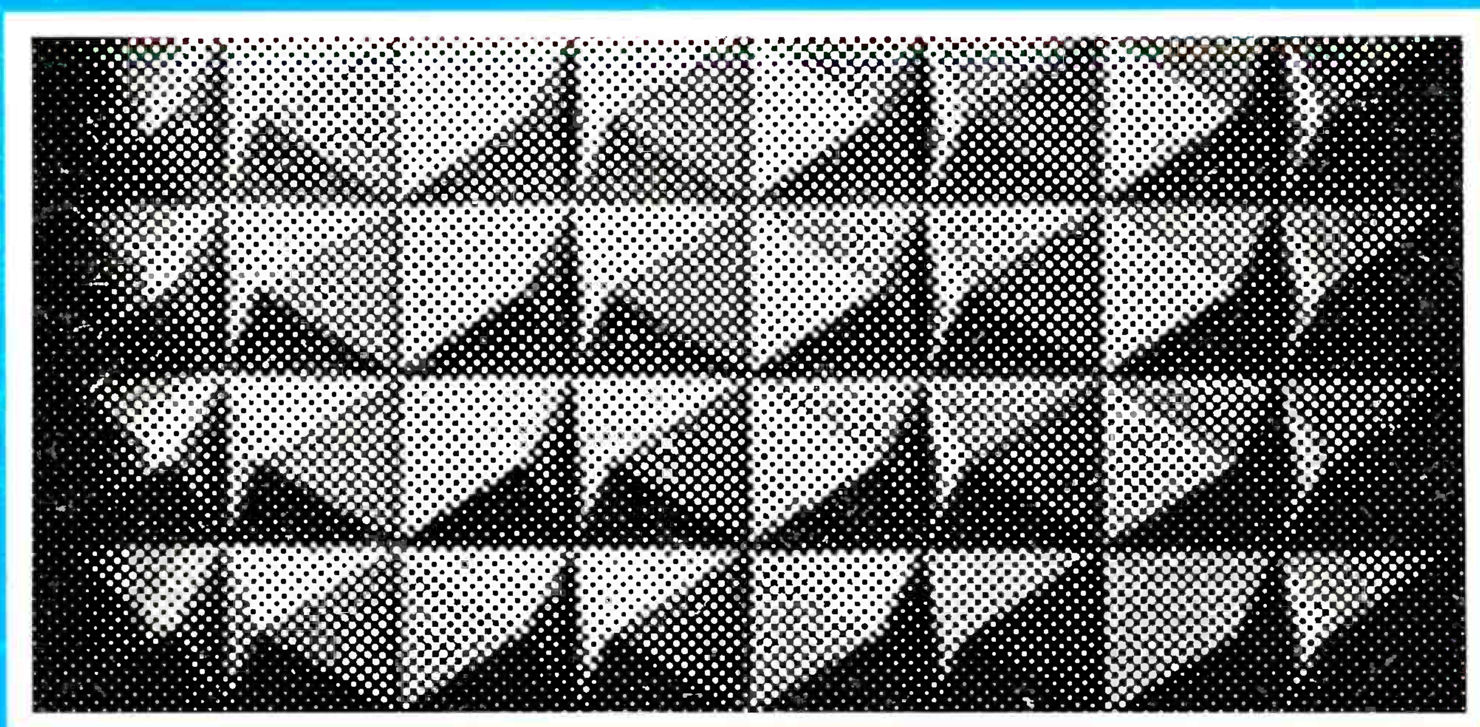

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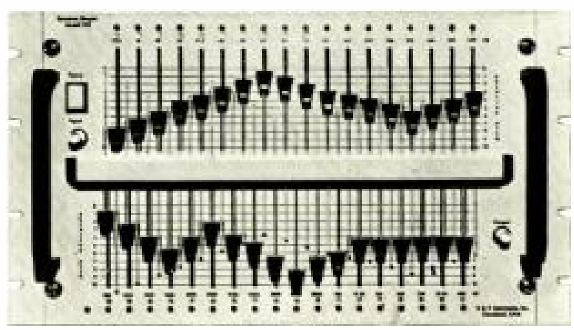

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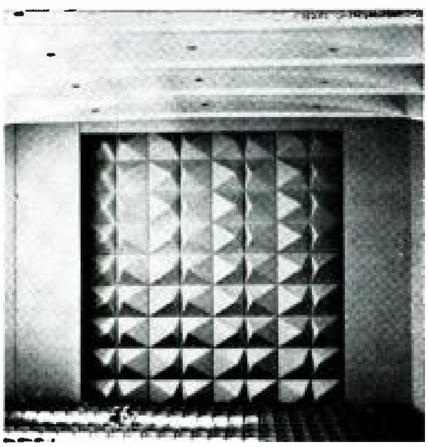
Coming Next Month

● April's **db** will contain floor plans, schedules, and the papers to be given at the 34th National Convention of the AES to be held in Los Angeles at the end of April. This will provide an excellent in-hand guide to every aspect of the Convention.

The promised article by Leon Wortman on Ampex' theater sound systems will be featured.

In the how-to category, assistant editor Richard L. Lerner has prepared a detailed account of the small-scale preparation and construction of printed-circuit boards.

About the Cover



● This stylized visualization is acoustic treatment mounted at the rear of an intimate theater that is a part of the Oberlin College Conservatory of Music in Oberlin, Ohio. See David L. Klepper's article on Architectural Acoustics beginning on page 18.

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Sound with Images

MARTIN DICKSTEIN

This new column is devoted to the field of audio-visuals. Martin Dickstein has had considerable practical and theoretical experience in the design and installation of systems. He has written for several technical publications including two earlier appearances in **db.**

● In the audio profession, field specialization has become a reality. Some people are experts in the recording industry as equipment designers, others are mixing engineers, tape editors, disc cutters, or other divisions of this field. Among broadcast engineers there are many who specialize in tape editing, remote program control, etc. In the public address/sound reinforcement branch of audio, there are also equipment designers, systems engineers, control operators, and acoustic specialists (to name just a few of the unique divisions). All of these men, although their interests and knowledge may be greatly diversified and their work may overlap two or more fields, have become or are fast becoming specialists in their own line.

However, it has become recently evident that for one in the audio profession to do his job really well, he must be aware of advances and developments in other phases of this rapidly expanding field. His knowledge must invade all the closely allied fields of informational dissemination if his usefulness is to continue to grow.

Along with the improvements and advancements in audio there have been similar changes in the visual fields. The desire to "get the message across" better or stronger has caused a tremendous expansion in the field of image projection. No longer is a slide projector merely a projector of slides. A film projector can be modified in a number of ways to provide the image required for a particular need. Projection systems do not necessarily consist of an "image thrower" and a screen. New concepts in the fields of mass entertainment, informational dissemination and education, to name a few, call for further development in the art of motion picture and slide making as well as their presentation to the audience.

In the not too distant past the producers of visual shows learned to rely on the knowledge of an audio professional to enhance the over-all impact of what might otherwise be just another movie or slide show. Very gradually, the two fields began to work more closely together and, as was demonstrated at most of the Expo 67 presentations, the limit of this marriage has not yet been reached.

One positive fact was illustrated by the most popular exhibits, that obtaining the greatest total effect upon the public depends on the proper blending of the visual art with sound. One without the other could never have achieved the same results. This mating of the two media has created the need for a man with both technical knowledge and a sense of the artistic. It proved that such a man, an *audio-visual specialist,* had to be part of the production team to smoothly join the ideas of the creators with the know-how of the equipment engineers. It provided concrete evidence of the necessity for the specialists in both audio and visual fields to know something of the other fellow's work so that each could better perform his own function in the over-all program.

This column will try to give the audio specialist a broader knowledge of the visual fields. From time to time there will be discussions of projection techniques, lens requirements, projector modifications, optics, tricks of image projection, lighting and meeting room design for visual presentation, charts and formulae, references for further investigation, use of closed-circuit television, rear projection, and other subjects of interest.

This column has been introduced to assist *you;* thus it will depend a great deal on you. If, in your work or just out of interest, you come up with a problem in the visual field, let us know. Perhaps this column can supply the answer. By contributing your thoughts, opinions, knowledge and experience you may help someone else, who in turn can do the same for you.

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The Feedback Loop

JOHN A. McCULLOCH

● Designs and ideas on console construction are many, and while there is a basic console plan, any design which permits the flexibility desired for the particular application without becoming unnecessarily complex is satisfactory. At present a console is being constructed *not* primarily for use in recording mixing, but rather for the investigation of miking techniques, and the microphones themselves. In order to provide maximum flexibility and usefulness, expansions of some circuits not usually found on general consoles were incorporated into this unit. Perhaps the circuits as used in this console are not directly usable, but could be adapted to your particular design.

Since the module unit chosen for the input channels has separate high- and low-level inputs (selected by a switch), the additional wiring required was utilized to perform other functions. Each input pair contains a phase-reversing switch, providing control of microphone and high-level phases when mixing multiple sources into a single channel.

After the phasing networks there is a dpdt switch to place the high-level pair in parallel with the nomal microphone pair. Insofar as the combining is after the phasing, the application not only permits the addition of two microphones before any amplification, but they may be combined in-phase, out-of-phase, or in-phase themselves but out-phased with respect to the remainder of the microphones used in the pickup.

A fourth switch would permit the selection of the combining action to be either in parallel as shown in FIGURE 1, or in series as shown in FIGURE 2. Continuing along the input circuitry—a pair of test points dropped from the microphone, just before amplification, are brought up to a panel. The purpose here, is to provide a means for measuring, with a calibrated, wide-range dB meter, the actual output level of the microphone. The meter should have calibration means for the standard impedances used in microphones as well as the normal 600 ohms. Some manufacturers of amplification equipment claim a margin of 10 dB between normal operating level and maximum capabilities of the amplifier (both input and output levels should be considered) to be sufficient. I prefer to plan for a minimum of 16 dB, and several recording engineers, whose work is constantly at the forefront, claim a requirement of from 24 to 26 dB.

Whichever margin you feel sufficient for your operations, the use of these test points, the wide-range meter and an input pad (either in-console or in-line plug-in type) make it possible to accurately control the input level to the first stage of the amplifier to obtain the best possible conditions for high-quality recording.

Test points at other stages might be considered, but if the design of the console was correct and conservative, the input is the main variable. Certainly it is the most critical. We desire the highest input level possible to avoid a poor signal-to-noise ratio. Yet, we must provide a good peak margin to prevent any possibility of overload. Measure and know the knee of distortion for your particular input stages (the point on the distortion *versus* power curve where the distortion first starts to rise rapidly). Use this point, not the actual point of observed clipping, as the maximum input to the amplifier. Subtracting your peak margin will give the normal operational input level.

It is common to provide echo feeds both before the fader (pre-echo) and after the fader (post-echo). The question arises: should this post-echo be tapped directly after the fader, or after equalization? Perhaps a third name-designated position (eq-echo?) should be adopted to clarify just where the tap *is* made. The versatility of all three positions should be evident, but the complexity of switching may not be to your taste or pocketbook.

At the output of the unit or channel, we normally find buss, submaster, or master channel selection. Bussing or

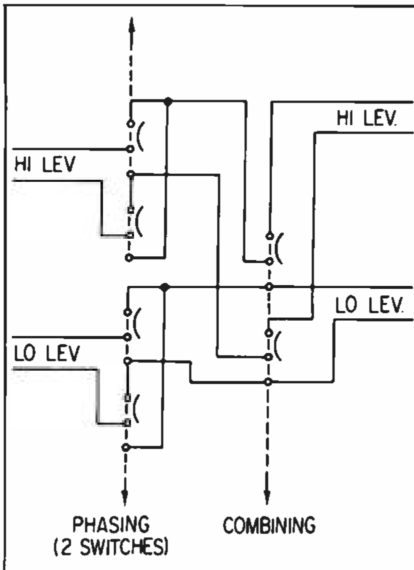

Figure 1. This switch permits the combining action to be in parallel.

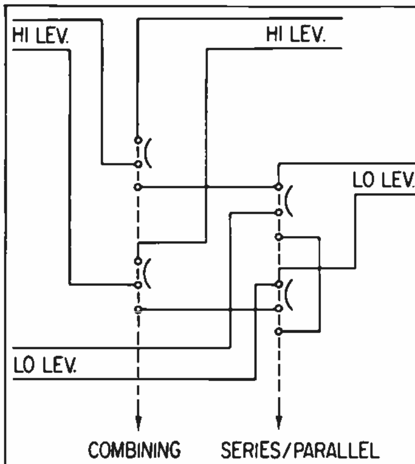

Figure 2. With this configuration combining action is in series.

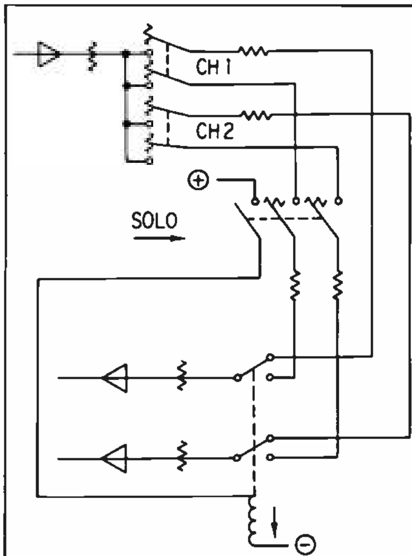

Figure 3. Selecting solo busses. In this single-line illustration, the switches are on terminating resistors.

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combining networks are standard items, whether they be passive or active. As I prefer to have the phasing ability at this point, as well as at the input, the console uses a balanced buss of the "O" configuration. A separate key is also available which disables the regular busses, and—by relay—substitutes a set of busses called *solo*. (See FIGURE 3.) Because of the switching design, this solo position places the selected channel alone on the output, in its exact level and distribution among the output channels, as it would appear in the final mix. It is possible to achieve this effect without affecting the normal recorder outputs by moving the activating relays to the inputs of the control room monitoring system. This would then necessitate the addition of duplicate amplifiers and dual tracking faders in the output sections of the console.

The ability to hear a single performer or section is an added benfit to the mixer and producer during the pre-recording balancing. But whether this facility would be desired during the take, without disturbing the take, is again a matter of complexity and cost.

Under normal conditions the console's master channel faders are just that, but for special effects, or original mastering on monophonic/two channel, each of the four output channels has the added capability of being switched to any or all of the recorder busses. Thus the masters in effect become sub-masters. A final master could be added, to be effective only when the mixer desires the additional control. At present a board fade is not difficult as only four channels are present. Should expansion to eight occur, then the physical requirements would dictate a final master.

Finally, keys that are set for each recording machine permit deletion of the normal bussing, and substitution of the recorder into the master channels for playback and duplication/dub-down. The keys also disconnect the record feeds to that machine (to prevent a loop path, should the machine's monitors be left in *record* position). The *normal* play route for duplication and dub-down is through the high-level inputs on the input module. This routing permits the addition of additional equalization, echo, or other control conditioning that may be desired.

Ideas such as those above, and others incorporated in the console have been gleaned from exceptional consoles around the country, as well as designed for the special requirements of this application. It is desirable that other ideas that you may have devised, whether in this area or any other, be disseminated among others in the audio field. THE FEEDBACK LOOP is your sounding board for ideas, and your fact sheet for co-operation information.

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Theory and Practice

NORMAN H. CROWHURST

Norman H. Crowhurst, P.E. is a prolific writer on all things electronic. His byline has appeared in nearly every periodical on electronics and he has authored many textbooks on engineering and mathematics. His distinguished career as an engineer spans several decades. An authority on both sides of the theory and practice fence, Mr. Crowhurst has both the insight and the credentials to bridge the gap between these two poles.

● Much of what I have written over the years has concerned the relationship between theory and practice. A common viewpoint regards theory and practice as mutually irreconcilable: theory is one thing, practice another. The man of theory has his head in the clouds, the man of practice does what he knows from experience will work.

Of course, recent scientific development has served to prove otherwise, to an extent. The best known example is the atom bomb: theory developed this new weapon which, when made, "worked." The transistor and many electronic devices are examples less widely known. But in every-day electronics, including audio, we have countless examples where theory and practice still appear to be contradictory to many of us.

It is in these examples that this column will take an interest. Actually, *true* theory is never wrong. *Incomplete* theory is what practice often proves wrong. Disregarding details thought not to be important leads to an erroneous result; the thing thought negligible wasn't!

Articles and chapters I've written on matching have brought responses from theorists, telling me that my references to standard definitions are incorrect, particularly regarding the quantity called *insertion gain*. They infer that my practical slant is at variance with EIA and IEEE definitions.

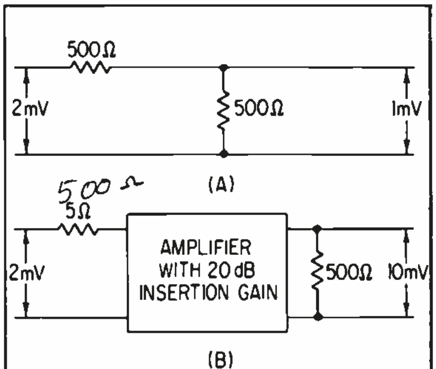

Figure 1. The definition of "insertion gain": (A) transfer between two impedances; (B) gain when amplifier is inserted between them.

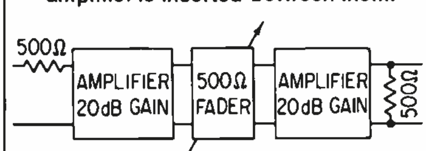

Figure 2. The situation explored in this column to show various deviations that can occur.

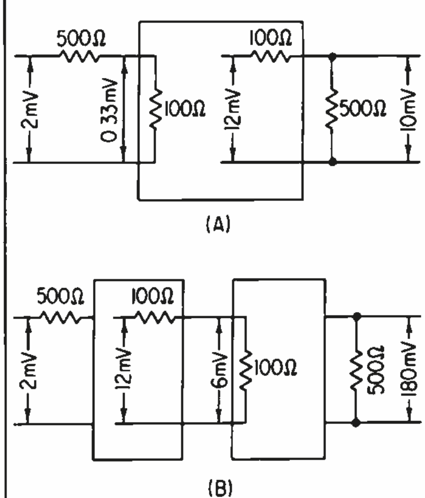

Figure 3. One deviation: both internal impedances are one fifth nominal. (A) what this means in terminal-input and open-circuit output volts; (B) the cascaded combination, showing how 5 dB extra gain appears.

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For several years I was a member of the committees of these institutions that produced the now-accepted definitions of insertion gain. A lot of time was spent around the table, trying to make the wording more meaningful and "foolproof." We believe we did a good job.

This problem, like most theory and practice dilemmas, can best be understood by example. We'll assume that a unit we're going to use in a variety of circuit positions is a line amplifier with an insertion gain of 20 dB. Input and output impedance are both 500 ohms. By definition, this means that inserting this amplifier between the source and load impedance, each of 500 ohms, will raise the output level by 20 dB, or 10 times.

Without the amplifier in circuit (FIGURE 1A), a 2 mV input through the source impedance of 500 ohms will produce 1 mV across the output load of 500 ohms. Put the amplifier between them (FIGURE 1B) and the output at 500 ohms will now get 10 mV. By definition, and in fact, this is an insertion gain of 20 dB.

We naturally assume that the input is still 1 mV. But it may not be. The specified method of measurement requires that the voltage at the input end of the 500-ohm input resistor still be 2 mV. Whether the input to the amplifier is still 1 mV depends on whether, as well as working from a 500-ohm source, its own internal input impedance is 500 ohms.

But isn't this the same thing? No, it isn't! The rated input and output impedance of amplifiers refers to the impedances to which they should be connected, which may or may not be the same as the measured internal impedances of the amplifier itself.

In a tube amplifier, the input is a grid. To achieve a line-impedance input, a transformer is used that will give as much step-up as feasible from the 500-ohm source impedance, consistent with a specified frequency response requirement. The internal impedance, reflected back to the transformer primary, is unlikely to be precisely 500 ohms. It may be much higher, at most frequencies.

In a transistor amplifier, the impedance at the input is more readily controllable than in a tube amplifier, but this does not guarantee that the designer chose values to achieve precisely 500 ohms as input loading, as well as achieving satisfactory operation *from* 500 ohms.

The output is even more likely to possess this difference. Voltage feedback is generally used, operative at its nominal value when the nominal output load is connected. From this, the facts of life make the source impedance at the output much lower than the load impedance. On the other hand, if current feedback instead of the voltage variety is used, the source impedance will be higher than load impedance.

At the output end, the only way to achieve correct matching, in the academic sense, is to divide the feedback between voltage and current forms, so the resulting impedance agrees with the

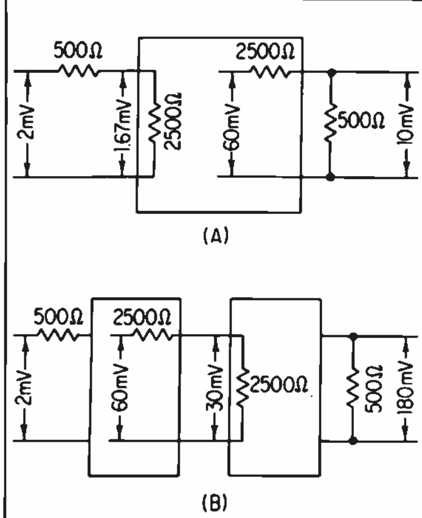

Figure 4. Another deviation: both internal impedances 5 times nominal. (A) effect on one amplifier; (B) over-all effect.

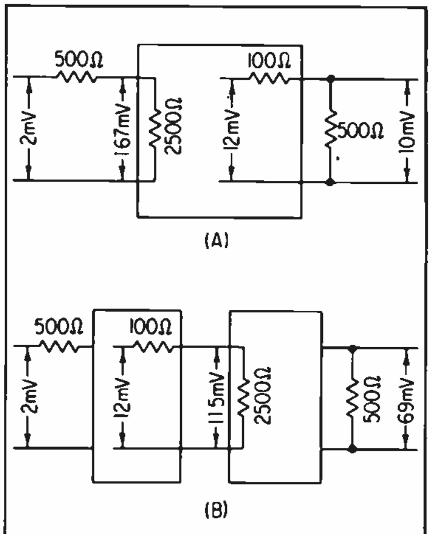

Figure 5. A third deviation: input 5 times, output one fifth nominal. (A) effect on one amplifier; (B) over-all effect.

WOW & FLUTTER METERS

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Besides the normal measuring connections on the front of the instruments, the back is provided with a connector for testing home tape recorders and record players. Other connectors are provided for the connection of external filters, oscilloscopes and graphic recorders.

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Input impedance: .... >10 kOhm.

Measuring ranges: ME-101 — ±0.02 to ±0.5% and ±0.1 to ±2.5%; ME-102 — ±0.15% and ±0.75%.

Frequency response of fluctuation indication: Linear position, 0.50 to 500 Hz (-3 dB points). Weighted position, according to CCIR standard with external filter as required.

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nominal load. Many amplifiers in service were not designed that way!

Let's use some figures to see what this means. Assume we cascade two of these amplifiers, with a professional gain control (fader) between them (FIGURE 2.) The input and output impedance of each of these line amplifiers may differ from the nominal load of 500 ohms for which they are designed. Let's assume each internal impedance may be either 100 ohms or 2,500 ohms.

At input and output of the whole combination, where we have shown true 500-ohm termination, it doesn't matter what the internal impedances are, because any difference from nominal is taken into account in the way insertion gain is measured and specified. But the connection between the two amplifiers can be seriously invalidated, especially when all attenuation is removed by running the fader all the way up.

Let's continue the assumption that the input is 1 mV (or to be precise, 2 mV through the 500-ohm input resistor): if all impedances have their nominal value, the output of the first amplifier, connected to the input of the second, will be 10 mV, and the output of the second will be 100 mV.

Now suppose the internal impedances both at inputs and outputs are 100 ohms. The first amplifier, instead of being loaded with 500 ohms, is loaded with 100 ohms. Removing a true 500-ohm load would allow the output voltage to rise from 10 mV to 12 mV (FIGURE 3). Then connecting an actual load of 100 ohms, to match the internal value, will load its output down to 6 mV.

The 20 dB gain of the second amplifier (to give 100 mV output) assumed an input actually measuring 100 ohms was connected to a 500-ohm source, with an actual input voltage of 20 mV. So the voltage at the input terminals, to give 100 mV output, would be 3.33 mV. Now our actual input voltage is 6 mV. So the output voltage will be 100 mV multiplied by 6/3.33, or 180 mV. We have 5 dB more gain than we're supposed to have.

Next let's assume both input and output impedances are 2,500 ohms (FIGURE 4). The actual input voltage, instead of being 1 mV, will be 5/6 of 2 mV, or 1.67 mV. And the output voltage will be 60 mV open-circuit, to load down to 10 mV with 500 ohms. So 2,500 ohms will only load it down to 30 mV.

Had the source for the second amplifier been 500 ohms, the actual input voltage would have been 16.7 mV, to produce 100 mV output. So 30 mV input will produce (30/16.7 x 100=) 180-mV output; just as in the other case, this is 5 dB extra gain.

Finally, let's assume the output impedance is 100 ohms and the input impedance is 2,500 ohms (FIGURE 5). As in the second case, the actual input voltage will be 1.67 mV. The open-circuit output voltage will be 12 mV, as in the first case. The actual output voltage will be 25/26 times 12 mV, or 11.5 mV.

To get the nominal 100 mV output from the second amplifier, the input should be 16.7 mV. So 11.5 mV will give about 69 mV output. A loss of 3.2 dB from the nominal 100 mV.

Now let's think about what the fader will do. With about 20 dB attenuation, it will present close to its nominal impedance at both input and output, so the nominal condition is closely approached. Over-all gain will be close to 20+20—20=20 dB. But as attenuation is removed, the gain will change by the amounts calculated just now.

Where both input and output impedance err from nominal in the same direction (both high or both low) the last few studs of the fader will change gain by more than they're supposed to do. If the fader is designed to insert 1.5 dB per stud, it will more likely become 3 dB or more on the last stud or two.

On the other hand, where input and output impedance err in opposite direction (input high, output low, or *vice versa*) the last few studs will seem to be almost inactive. The 1.5 dB per step nominal may drop to less than 0.5 dB.

Does that explain some unpredicted gain problems you may have encountered at some time? That discrepancy is the simplest consequence of this effect. Next month, we will discuss other consequences, such as changes in frequency response or increase in distortion that can come about at the same time.

Meanwhile, please don't write to tell me that I'm misjudging the standard method of specifying insertion gain. Or that manufacturers' ratings are deceptive when their internal input and output impedances don't match their nominal values. The problem arises because we have so many variables that it's difficult to specify them all without causing confusion. The only solution is a little better understanding of the theory!

Beyond this, I have several ideas of my own to discuss in future columns. But many of the ideas are going to come from you, the readers. It takes a couple of months between the time I work the idea up and you read it in print. So start the questions coming if you want to see yours answered soon!

Letters

The Editor:

Your editorial in the January issue of **db** interested me and brought back memories. I realized that when we moved to this country (in 1953) one of the "translations" I had to make (like *tube* for *valve* and *wrench* for *spanner*, not to mention *sidewalk* for *pavement*), was *public address* for *sound reinforcement.*

Yes, as early as 1933 I was employed by a British company that specialized in two kinds of service: sound reinforcement and sound relay. Each had its own problems. Sound reinforcement intensified sound in the same room. Relay carried sound into other rooms and buildings.

Back in those days there were *lousy* installations, but the cause of the lousiness was usually failure to accommodate to the acoustic problems. The British would never have tolerated the raspy quality that Americans accepted in the name of public address.

At that time I could not have said that we British never used public address, only sound reinforcement. Americans wouldn't have known what I meant. Few had heard sound reinforcement. To try to explain (as you did in your editorials) would have sounded so dreadfully snobbish. But eventually the viewpoint has changed. Now, here too, the public will no longer accept high distortion.

I don't remember the date when St. Paul's Cathedral first installed line radiators with a time-delay system, to overcome the building's appalling reverberation problem, but it was before we emigrated. That was a problem triumphantly solved by sound reinforcement.

However, I suppose the main reason I could not have insisted on use of the term sound reinforcement then was the number of operators, scattered across this country, who knew nothing better than public address, as you so succinctly describe the distinction. And **db** would have been completely beyond them!

Norman H. Crowhurst, P.E.

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Sound Re-inforce-ment

PHILIP C. ERHORN

● In last month's discussion of the large auditorium sound system, in among the many faults of the system were some good factors. One was the professional approach used in designing a conventional low-impedance mixer, such as we use in broadcasting and recording. The great majority, of "p.a." systems are assembled from packaged mixers which at best afford several microphone inputs and a line or a power amplifier output. Some of these packaged mixers include a "tone control", a feature with large potential danger from the standpoint of feedback or muffled speech articulation, where hi-end roll-off is used in an attempt to alleviate feedback.

However, when a conventional 600-ohm type mixer is used in the system design, then valuable tools such as sharp cut-off filters and graphic equalizers may be easily inserted. These tools have existed for many years and are in widespread use in studios and other professional systems. Unfortunately, they do not adapt to hi-Z mixers and have been largely neglected by price-conscious sound system installers, who operate in a competitive field.

Most large public spaces have several points in their acoustic characteristics capable of triggering feedback, and because of their large size and thermo-acoustic effects, they will be naturally bass-heavy, thus needing some assist in improving speech articulation. Generally, compensation can be made by either sharp attenuation of system bass response below about 100 hertz with a filter, or the more widespread but gentler roll-off with an equalizer. Judicious equalization with a graphic type device may be sufficient to both alleviate feedback and improve articulation, since perhaps a half dozen or more small sections of the audio spectrum may be either boosted or attenuated simultaneously. More difficult situations, demonstrating sharp acoustic peaks, must be handled with either a tunable dip filter or one or more fixed dip filters. Reference to the **Journal of the Audio Engineering Society** for the past couple of years will provide good reading on this subject.

It will also point up the fact that architects rarely call upon a qualified acoustic consultant when designing large public areas of the enclosed type. The acoustic and/or sound system expert is called in on these jobs only *after* the damage has been done.

In addition to auditoriums, there are notably airport and railroad terminals which seem to suffer universally from acoustic and sound-system problems. In defense of the problems of the architect, I must point out that his client may frequently insist upon the preservation of visual features, to the detriment of acoustical details. It is typical of airport and railroad terminals to claim that acoustic compensation has been attended to by the expedient of cementing acoustic tile to the ceiling areas, or at best by means of a hung tile ceiling, while walls and floors remain hard, reflective surfaces. Low-frequency absorption is negligible and echoes are only a part of the remaining problem with which any sound system must then cope.

In any attempt to solve the potential problems of the sound-system designer, as related to a host of typical architectural restrictions, we must be concerned with microphone and loudspeaker characteristics – particularly polar patterns, mixing facilities and associated filters and equalizers, and audio power distribution with proper impedance matching. In order to avoid oversimplification of this thumbnail list, let's look into each area separately, starting with a professional mixer approach.

FIGURE 1 illustrates a somewhat abbreviated system diagram for a mono system made up of typical professional components. A typical physical layout of the mixer portion is pictured in FIGURE 2. Although small, this system has excellent versatility and operating convenience. Its integral little jackfield affords access to the appropriate points in the system for insertion of filters and equalizers. These devices may be patched in during initial set-up testing, and either left on a patchable basis or wired in permanently later.

Levels as shown are typical for the system, and include provision for the insertion loss of a passive graphic equalizer, plus a nominal gain reduction of at least 10 dB through the use of a compact LDR-type compressor. If a lossless equalizer is to be used, it is possible to eliminate the booster amplifier, assuming slightly higher gains than those shown for the preamplifiers and the program amplifier. Several mic inputs, a hi-level input and a transcription input are typically indicated. Each input has an associated vertical attenuator, and all impedances excepting the 150-ohm microphone inputs, are 600 ohms. Amplifiers are plug-in transistor units with both input and output transformers to avoid common grounding problems.

The mixer shown in FIGURE 2 terminates in a line output of plus 4 dBm and, depending upon the particular need, a number of medium (or even low) power amplifiers are bridged across this line. In turn, the 600-ohm power output (or more typically, 70-volt output for higher power amplifiers) is fed to groups of speakers, which are bridged across a common speaker line with a line-to-voice coil transformer at each speaker.

Note that whenever compression is used in a sound-reinforcement system, the degree of gain reduction must be such that gain recovery during speech or music pauses, will not put the system into feedback.

The various types of filters which we find most useful in sound systems have no insertion loss from a system gain standpoint, and may be purchased in both fixed or variable hi-pass or lo-pass types, and as notch filters. Notch depth is about 14 dB with a quite sharp notch width. Attenuation rate for the hi- and lo-pass units is nominally 18-dB per octave, but may be specially ordered at a 30-dB rate. These filters may be inserted in individual input channels; in individual power amplifier channels; or more logically, in the over-all program channel.

Program-type equalizers are relatively useless in sound-system work, since their curves are based upon a fixed hinge point, and they are so broad in character that they may actually help to cause, rather than eliminate feedback, as with a tone control. Where an equalizer is indicated, the versatile graphic, (FIGURE 3) with multiple adjustable frequencies of a mirror-image type, is most appropriate. Such an equalizer may be used to put a hole in an area of feedback potential, or give

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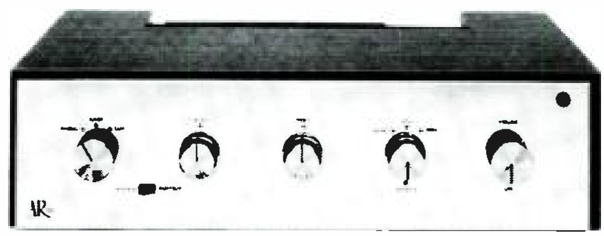

The AR amplifier delivers 60 watts per channel continuous output at less than 0.5% harmonic distortion, 20 to 20,000 Hz, both channels operating simultaneously into 4 ohms; 50 watts per channel into 8 ohms.

One of the most important specifications of an amplifier is its power output. In view of this, consumers might expect this measurement to be presented clearly and accurately in amplifier advertising. This has not been the case. In recent years, a variety of vague or irrelevant terms has been used by manufacturers to describe power output: music power, solid-state power, stereo power, audio power, transient power, transistor power, IHF power and others. The list includes terms invented by manufacturers and applied to their products alone, as well as standards of measurement known only to advertising copy writers.

Acoustic Research uses the definition of a watt given in physics texts: work done at the rate of 0.7375 ft.-lb./second. We know of no "transient watt" or "music watt" which science recognizes. AR amplifiers are rated exactly as we measure them, with both channels continuously delivering at least the rated power without exceeding our harmonic distortion limit of 0.5%, or the I.M. distortion limit of 0.25%. The laws of physics and the nature of music require that power measurements, if they are to be meaningful, be made with a steady, uninterrupted tone, similar to the purest sound of a pipe organ. AR amplifiers must deliver their rated power at all frequencies to which the ear responds, not just at 1,000 Hz, where most amplifiers can deliver much more power than at the extremes of the range of hearing. Distortion measurements are made through the AR amplifier's phonograph input because music must go through the amplifier this way—even though performance might be better without the preamplifier in the circuit.

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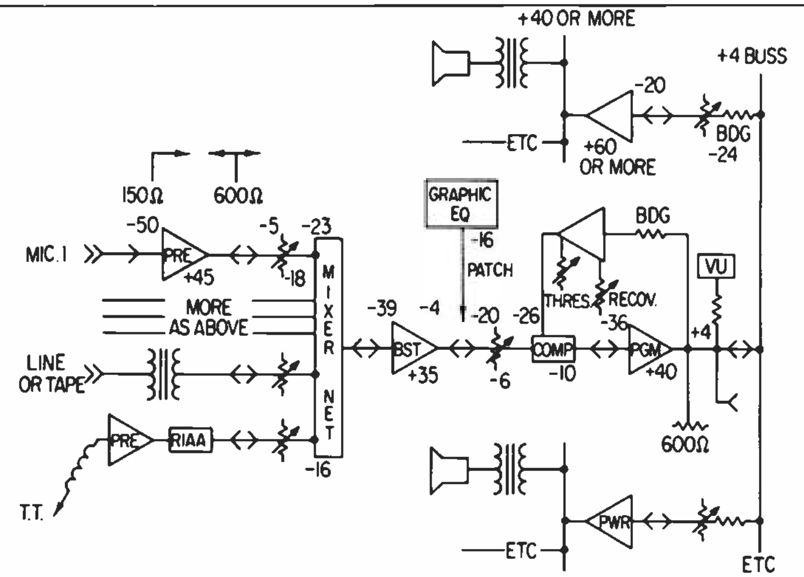

Figure 1. An abbreviated block diagram of a small sound-reinforcement system.

Figure 2. The physical layout of the mixer portion of a small sound-reinforcement system.

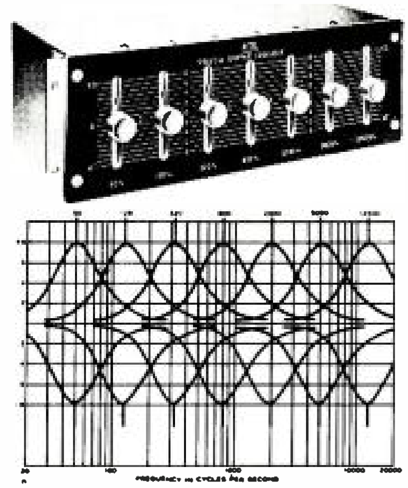

Figure 3. In (A) an Altec graphic equalizer is shown. (B) shows the possible boost or cut attainable with the individually-controlled seven channels.

moderate boost in those regions where speech needs an assist to improve articulation.

These equalizers may be ordered with a specified number of adjustable channels, and at specified frequencies, if desired. They are available in both passive and active configurations. The diagrammed system provides sufficient gain to accommodate a passive type with an insertion loss of 16 dB. It is shown as being inserted between the booster and the master-gain control, and will of course affect all speaker locations.

Filters are logically patched into the program line at any point past the mixer network, again affecting the entire system. If it is desirable to filter an area fed by one group of speakers only, then the filters(s) would be patched into the jacks on the input to the appropriate power amplifier(s).

As you can now see, the conventional professional mixer approach has given us the facility to quickly try out and then permanently utilize whatever tools we may need to correct acoustical problems existing in a given room. In addition, the simple idea of foregoing the usual high-power amplifier, in favor of bridging a number of low- or medium-power amplifiers off the mixer line output, gives us area control of volume levels – the ability to compensate each area through filtering and equalizing – and ease of impedance matching. In the January issue of **db**, J. F. Walthier gave an adequate rundown on proper impedance matching by the constant-impedance or the constant-voltage approach. Either method will eliminate power wasted in heating improperly matched lines, as well as affording a degree of level control for each speaker.

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The Audio Engineer's Handbook

GEORGE ALEXANDROVICH

The Lathe

● Of the several types of recording lathe in use today, some are no longer available new. Only a few manufacturers continue to offer new equipment, the most popular and sophisticated of which are by Scully of Bridgeport, Connecticut and Neumann of Berlin, West Germany. Both are excellent systems, offering the user a high degree of automation.

Even with the best lathes, you must know how to install a cutter properly and how to operate it. Regardless of the recording system, cutter type or lathe, certain basic tests must be performed after installation of the recording chain.

The lathe itself must be at least as good as the best playback turntables in rumble, speed constancy, and accuracy. Flatness of the platter, bearing noise, operation of the feed screw and main shaft play (the one holding the turntable), are all points of concern. Rigid standards of performance must be met for the functioning of the pitch selector mechanism. The feed screw must provide a pattern-free cut.

Testing the motion of the platter requires techniques similar to ones used in evaluating playback equipment. A flutter bridge is an invaluable tool if you are fortunate enough to have access to one. A proper signal as required by the bridge (around 3 kHz for most, there is a standard at 3150 kHz) is recorded on disc and then played back on the lathe itself. Play the disc several times, each time moving the disc in relation to the platter to check for the worst possible condition where recording and playback flutter indications are additive. This indication should be divided by two to disregard the speed error of playback.

I prefer this method of testing for flutter over the use of standard recorded test records. The test records suffer from the inherent flutter of the machine on which they were recorded; thus I feel this method to be less accurate.

Flutter can also be indicated without a bridge (albeit less conveniently). A generator and oscilloscope are necessary. With the 'scope set for identical display of vertical and horizontal inputs, a lissajous figure will be seen. This will be a slanted line when the horizontal and vertical inputs are identical. When one of the signals feeding the 'scope varies in frequency (the signal from the playback equipment) a display ranging from a slanted ellipse to a circle will be produced, depending on the phase shift between the two signals.

Using this technique for flutter measurements demands an understanding of what accuracy must be achieved. Speed constancy specifications for turntables are often quoted at 0.2 per cent maximum deviation from the nominal speed. Therefore, we should be seeking a maximum deviation at the recording lathe of 0.1 per cent.

With this in mind, some thought should be given to the selection of a suitable frequency to display a usable pattern. With the usual flutter measurement frequency of around 3 kHz, a frequency change of ±3 Hz would indicate the maximum allowable speed fluctuation. Since the most common type of flutter is once per revolution of the platter or about 1.8 sec. per cycle, our measurements using 3 kHz would be impossible to interpret. (This is because 3 Hz is 1080 degrees of phase shift while a 'scope rotates its lissajous figure every 360 degrees.)

If instead we select a lower frequency, the speed deviation of the lathe can be expressed in degrees of phase shift per revolution, not exceeding 90 degrees in 1.8 sec. This is approximately 50 degrees/sec. or 1/7 of a cycle/sec. If we want to be able to see 0.1 per cent of the nominal frequency, then this frequency should be approximately 140 Hz.

(If 50 degrees is 0.1 per cent then 100 per cent is 50,000 degrees. Frequency is 50,000:360 or 138.888 Hz—close enough to 140 Hz for practical purposes.)

This means that the maximum deviation of the 'scope's lissajous figure from a line can be 90 degrees of swing between a line and a circle and back again to a line. It may be necessary to trim the oscillator settings slightly to compensate for generator frequency drift or lathe speed variation. The phase relationship between the two signals should always produce a slanted line. Double check the 'scope by feeding the same signal to both inputs so as to obtain the proper presentation.

The speed at which the phase of the playback signal deviates from the nominal frequency is the frequency of flutter. To determine the exact phase difference from the 'scope, refer to FIGURE 1.

Cutter mounting must be done with sufficient precision so that the specified angles are not more than 2 degrees off. (Two degrees sets the limit of separation between channels to 20 dB.) The first of several critical mounting dimensions is azimuth alignment. Looking from the front of the lathe, this is the normal (perpendicular) line to the surface of the lacquer. The cutting stylus should be parallel to this line. If the stylus position creates an angle greater than ±2 degrees off 90 degrees, all attempts to achieve more than 10-15 dB separation will be futile.

(The assumption here is that the subsequent playback stylus is at 90 degrees to the disc. If both cutting and playback styli are tilted to the same degree, separation will be satisfactory. It therefore becomes important to use a carefully mounted and calibrated playback system for measurement purposes — as well as visual examination of a cut groove containing modulation applied to one channel only. *Ideal* separation produces a groove with one ridge a straight line while the other carries the modulation.)

The next vital consideration of cutter mounting is the actual orientation of the cutting stylus. Most cutters today use either tapered-shank or bare sapphire styli. In the first case you must rely on your skill for proper stylus orientation, while in the second you depend on the accuracy of the clamping device as set by the cutter manufacturer.

There are several ways to correctly position a tapered-shank stylus. Most engineers use a tweezers to grip the shank of the stylus, plus some means of optical magnification to see the cutting edge of the sapphire. I've observed some younger engineers working without magnification.

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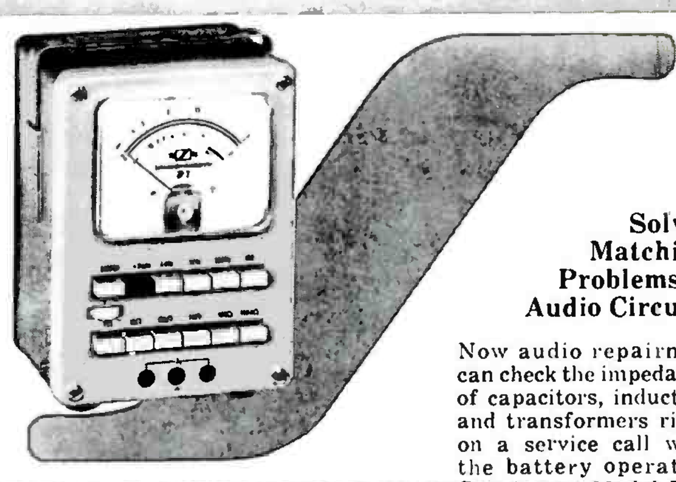

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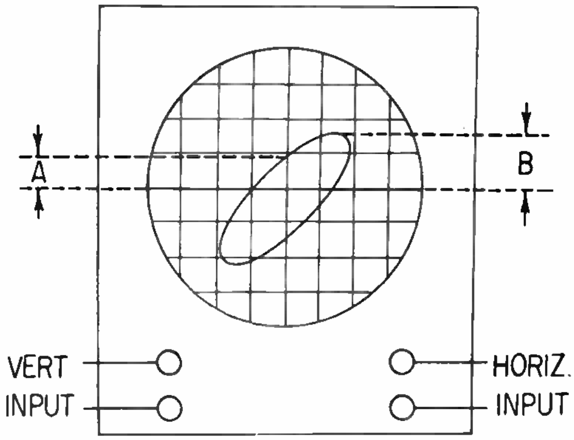

Figure 1. Use a 'scope to measure flutter. The formula is $\frac{A}{B}=\sin\theta$ where θ is the phase angle. A is the height of the y axis of the 'scope trace. B is the highest point of the lissajous figure.

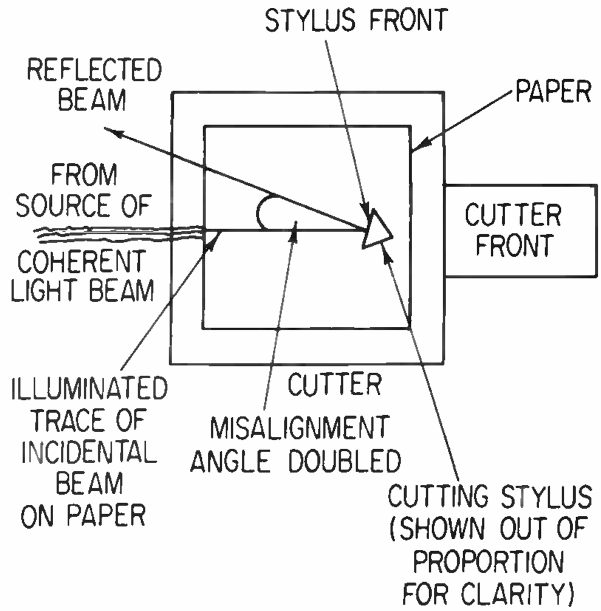

Figure 2. A coherent light beam coming from exactly 90° off where the flat of the stylus belongs will be deflected if the stylus is improperly oriented.

I prefer a method that is harder to use but far more accurate. It utilizes a narrow beam of light reflected off the surface of the sapphire's flat front. A narrow beam can be obtained from a focused flashlight or microscope light. Place a square of paper over the stylus tip so that the beam touches the paper indicating the direction of the source and the reflection. Rotate the stylus in its seat until both beams coincide. (See FIGURE 2.)

Since the angle of incidence is equal to the angle of reflection, the observed error is doubled. This makes alignment that much more accurate.

When a cutter is mounted particular attention must be given to the positioning of the leads to the head. In addition to the signal wires, there are heater wires supplying current to the stylus heater. There must be enough slack for them to flex without interfering with the vertical motion of the cutter. This also applies to the suction-pipe hose. It is recommended that all of these be fastened to the carriage itself so that regardless of the position of the cutter over the platter, the same vertical bias force is always present.

Quite often recording systems use an advance ball with the cutter to assure constant groove depth. The choice of using an advance ball or not depends on cutter construction, cutter cable flexibility, as well as the mechanical suspension of the cutter mount. When an advance ball is used, groove depth can be closely controlled because of the rigid coupling between the lacquer surface and the cutter body. A sapphire ball (actually a sapphire stud with a highly polished base) can accumulate considerable dust while it is riding the surface of a disc. This can cause lacquer smear and it can also change the actual groove depth if dust finds its way between the ball and the lacquer. Cleanliness, therefore, is of utmost importance if an advance ball is used.

If no advance ball is used, the depth of the groove is dependent on the lacquer resistance offered to the cutting edge of the stylus. This is a function of the lacquer hardness, stylus heat, recording speed, and the amount of suction. Suction affects the groove depth indirectly. Since the cutterhead is floating freely above the disc surface, positioned only by the tip of the stylus, every change in the amount of suction alters the vacuum or atmospheric pressure between the cutter and disc surface. However, the effect is partially offset by the stylus heat. This heater current is fairly constant so the actual heat at the stylus tip is dependent on the cooling effect of the suction. The more suction – the more cooling. The more cooling – the colder the stylus tip. The colder the stylus tip – the shallower the cut.

I suggest that an advance ball be used even with a free-floating cutter. Adjust it so that it rides 2-3 mils *above* the surface of the disc, without touching it. This will prevent the stylus from being broken if the cutter is dropped to the lacquer too fast.

Once the stylus is seated and adjusted properly, the heater wires must be dressed with a proper amount of slack to prevent biasing of the stylus movement at low frequencies.

(I once had to make a cross-country trip for the sole purpose – it turned out – of loosening heater wires which were causing a system to malfunction.) Let me hasten to point out that too much wire slack is also dangerous; wires may touch metal parts of the cutter and short out– with heater burn-out as the result.

Usually Nicrome V AWG 40 is used for heater windings. Wind four or five turns of wire around the sapphire and use a small amount of liquid ceramic to cement the wire in place.

Wire can be purchased from Industrial Heater in New York City and liquid ceramic from Sauereisen Corp. of Pittsburgh, Pa. The ceramic is available in paste form premixed, or in powder and liquid form ready to be mixed.

(To be continued next month.)

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Editorial

ACOUSTICS . . . SCIENCE OR ART? The article on Architectural Acoustics by David L. Klepper in this issue again raises this question. The construction of many music halls, all of which designs featured the work of acousticians, have been scored by public, press and audio professionals as unsatisfactory. One hall is too dry, another is too reverberant, a third has not enough intimacy. Fault will be found with any hall.

Does this mean that acousticians operate in the dark . . . that their results are produced by *experimentation* rather than *science*?

Not at all. Modern acoustics is very much a science. Very little that is done today is of the cut-and-try school. The acoustician involved in designing a hall or studio can create pretty much what is wanted. The core of the problem seems to be that the architects and underwriters of halls have broad ideas of what the room is to do. No one builds a specialized hall or studio; a variety of jobs must be accommodated.

A particular studio may be used mainly for small vocal groups, but since it might occasionally house eighty-piece orchestras, it must also be adapted for this need. A studio that fits both requirements can be built . . . but it will be involved and expensive. So the acoustician, against his better judgment and advice, compromises the room so that it is at least acceptable for both extremes. Then, the room is just that – a compromise. It is not highly satisfactory for either of our examples.

Thus the acoustician becomes a scapegoat for the studio's inadequacies, when in fact he could have created an acceptable situation had he been given a limited goal and a free hand.

There is a very real lesson to be gained from this. Many of you will be needing rooms that require specialized acoustic treatment. This should not be left to word-of-mouth chance. The acoustician is a proper part of the professional audio scene. His help and advice can multiply the usefulness of a room, thus increasing its potential for income. And this will offset many times the cost of acoustical consulting. *L. Z.*

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Architectural Acoustics

David L. Klepper

Certain ground rules must be followed to design a studio to specific acoustic needs. These practical facts suggest how to build or modify rooms so that the best can be obtained from them.

Part One: ROOM ACOUSTICS

THE professional worker in audio is involved with acoustics, regardless of his specialty within his field. Indeed, the fields of audio and acoustics overlap more than they are separate and distinct. The acoustics of recording and broadcast studios, theaters, concert halls, and other performing spaces affect the work of the recording and broadcast engineer—while the acoustics of any space requiring an electronic sound amplification system will affect the performance of the system and the work of the system designer, installer, and operator.

A theater, concert hall, auditorium, or a radio or television studio should be designed acoustically: first to form a good *room-acoustic* link between the live sources of speech and/or music, and the live listeners and/or microphone(s); and second to exclude noise. Both the room-acoustics and noise-control aspects of architectural acoustics have been under continuous development for many years. The goals of the room-acoustics design may be stated as:

- Insuring an even distribution of sound energy throughout the space.
- Controlling discrete echoes, rapidly repeating echoes (flutter-echoes) and bunching of the normal modes of vibration of the rooms which might emphasize certain frequencies.
- Providing the proper reverberation time characteristics.
- Assuring the proper ratio of reverberant-to-early sound, related to the shape of the reverberation decay curve and particularly important in spaces where live music is heard by live listeners.
- Providing a short enough initial time-delay gap, for early reflections, again important in *live-music* spaces.

Diffusivity and Echo Control

Even sound distribution and satisfactory control of echoes and normal modes is achieved where there are no extreme variations in sound pressure either as the frequency of the signal or the position in the room is varied. Diffusivity can be accomplished by avoidance of parallelism in the basic shape of the room, as in FIGURE 1, or large-scale modulations of basically rectangular shapes, as in FIGURE 2. The popular polycylindrical diffusers (FIGURE 3) are one form of "break-up" that can be added to basically rectangular-shaped rooms.

Where large, concave or flat surfaces are essential because of architectural or other planning considerations, echoes may be controlled by sound-absorbing treatment. Usually, however, diffusion by surface break-up and shaping should be considered first and the sound-absorbing treatment limited to that required for reverberation control.

Elements added to basically rectangular rooms for sound diffusion must be approximately equal or larger in scale compared with the wavelengths of sound energy being controlled. (The wavelength of a 100 Hz signal is 11.3 feet.) For speech studios a designer might restrict diffusion to the range above 200 Hz (5.65 ft.), but for music studios the range down to 50 Hz (22.6 ft.) or even lower – to the 20 Hz lower limit of human hearing – may be important.

In many auditoriums and concert halls, balconies or applied decoration in the form of statues, etc. are "natural" sound-diffusing elements.

Reverberation Time

We define reverberation time as the time required for sound to decay to one-millionth of its value (60 dB) after cessation of the sound source. In any real room it usually varies as a function of frequency and may be predicted accurately by the ratio of the volume of the room to the total sound-absorption present, according to the following formula[1]:

[1] This equation is called the Sabine equation after Wallace Clement Sabine, father of modern architectural acoustics. It appears generally applicable for must diffuse-large spaces, although the Norris Eyring equation or other modifications of Sabine's original equation are perhaps more appropriate for small or very dead rooms.

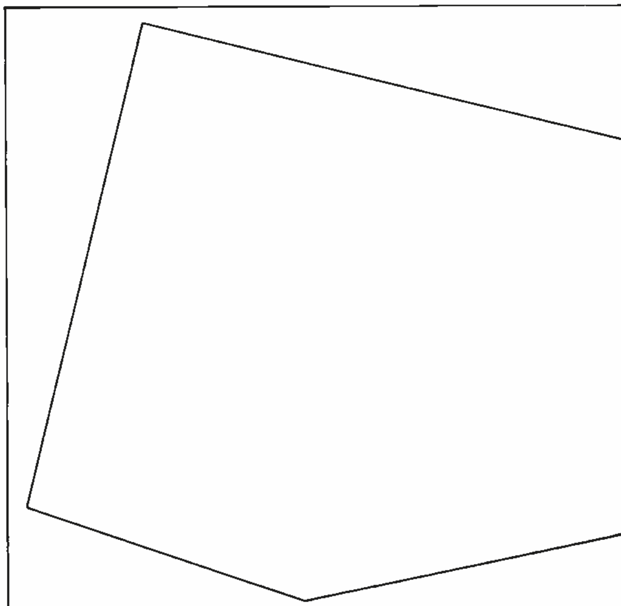

Figure 1. A non-rectangular plan for a studio.

David L. Klepper is associated with Bolt, Beranek, and Newman.

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$$R_t \text{ (reverberation time)} = \frac{0.049\ V}{S\alpha}$$

Where V = room volume
S = room surface area
α = average absorption coefficient
$S\alpha$ = total sound absorption

Although early studios were uniformly dead, there has been a trend from the early days of broadcasting to the present for more reverberant studios, particularly for music performance.

Beranek's suggested criteria for speech, general, and music studios still appear to be generally applicable and are illustrated in FIGURE 4. Large music studios should have much the same reverberation time characteristic as a fine concert hall. Speech studios should have essentially no audible reverberation with less or equal reverbation time at frequency extremes (low-frequencies and high-frequencies) as compared with mid-frequencies.

General studios may be a compromise between speech studios and music studios. Regarding auditoriums for "live" listeners, experience suggests the following mid-frequency reverberation times for different uses:

	Optimum	*Possible*[2]
Lectures	0.9 – 1.1	0.5 – 1.4
Drama (theaters)	0.9 – 1.4	0.5 – 1.6
Musical Comedy	1.2 – 1.5	1.0 – 1.7
Opera	1.5 – 1.8	1.4 – 2.0
Instrumental Recitals and Chamber Music	1.4 – 1.8	1.0 – 2.0
Orchestral Concerts	1.8 – 2.0	1.4 – 2.5
Vocal Recitals	1.5 – 1.8	1.4 – 2.0
Choral and Organ concerts (liturgical music – the upper limit depends on the type of music)	2.0 minimum	1.7 minimum

2 Good results possible provided other parameters are optimized.

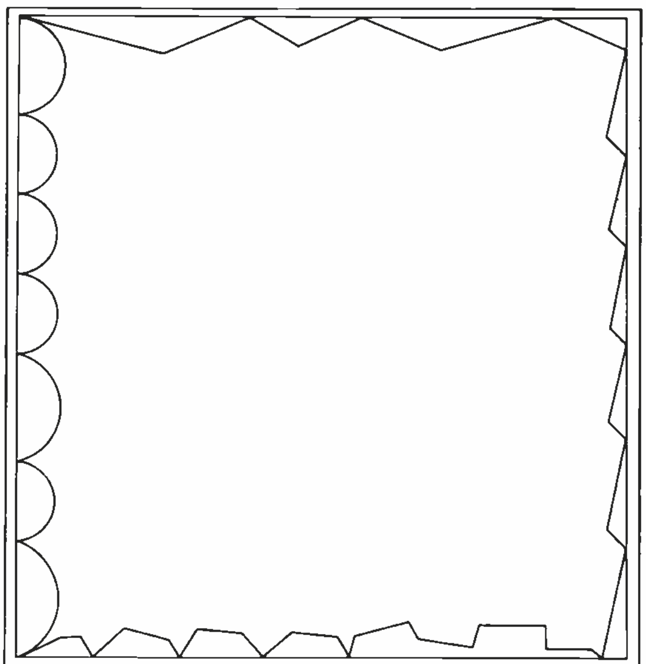

Figure 2. Sound diffusing wall modulations of several types applied to a rectangular room.

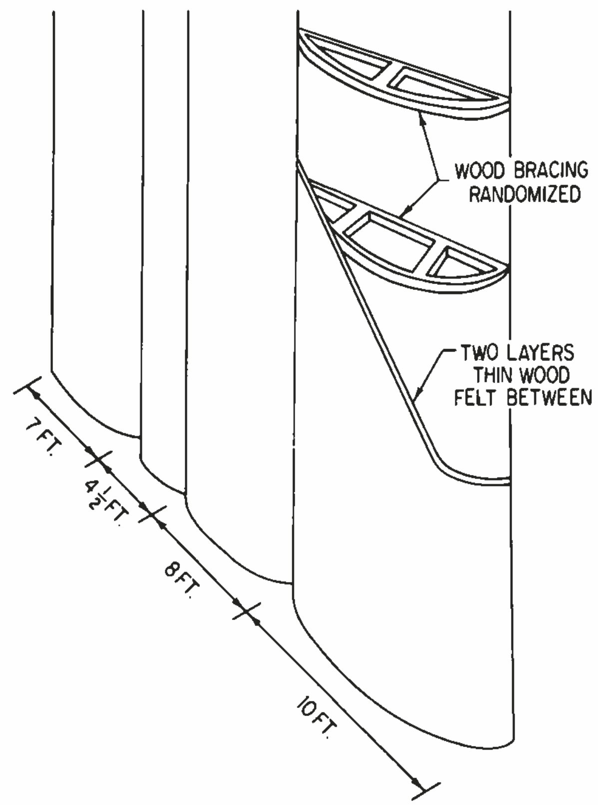

Figure 3. Typical polycylindrical diffuser construction. The diffusing effect of a cylinder is maximum when diameter equals one-half wavelength, and a combination of different radii will provide optimum diffusion over a wide range. The construction shown also provides low-frequency absorption.

Occasionally a reverberation time calculation will reveal a studio or auditorium on the "dead" side (too low a reverberation time), even without any applied sound-absorbing treatment. For example, the calculated reverberation time for a studio planned for music broadcasting may be under 1.5 seconds at mid-frequencies, even with all interior finish materials hard and sound-reflecting. The reason may be a relatively low cubic volume (interior volume less than say 40,000 cubic feet), with the predicted sound-absorption of the 100 members of a full symphony orchestra present.

Economics may dictate the design of studios or halls below optimum size. Under these conditions the wise designer or acoustical engineer will often refrain from attempting to make the hall or studio as live or reverberant as possible within the available volume, because under certain conditions in small halls or studios, he would be concerned that the space would simply be too loud for proper performance conditions. Instead, particularly in studio situations, he might choose to provide a relatively dry acoustical environment to be supplemented by electronic reverberation devices, either within the hall itself (for performers, listeners, *and* recordings) or for recordings alone.

We have been discussing mid-frequency reverberation time goals for various spaces. These are reverberation times measured in the 500 – 1000 Hz range. The variation in reverberation as a function of frequency is also very important; and it is a good measure of the "frequency response" of the room. For example, a concert hall with a reverberation time of 2.8 seconds at 125 Hz, 2.0 seconds at 500 and

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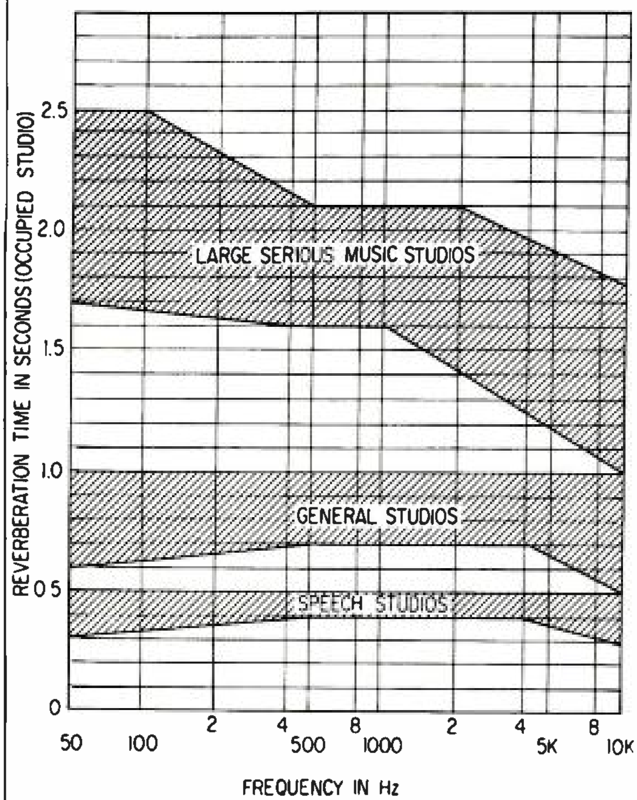

Figure 4. Suggested studio reverberation criteria.

1000 Hz and 1.6 seconds at 2000 Hz would be characterized as *warm*. On the other hand, a similar hall with reverberation times of 1.5 seconds, 2.0 seconds, and 2.0 seconds at 125, 500-1000, and 2000 Hz, respectively, would probably be characterized as *harsh*, or *bright*. Today, acoustical engineers plan large halls or studios for music performance to have rising low-frequency reverberation time characteristics: that is longer reverberation times at low frequencies than at middle or high frequencies.

In a speech studio, particularly a small one, a longer reverberation time characteristic at low frequencies would be characterized as a *boomy* acoustical environment. Speech-acoustics spaces should generally have flat reverberation time characteristics. Studios and auditoriums planned for both speech and music may be planned as a compromise, or — as discussed earlier — adjustable treatment or electronics may be employed to bridge the gap between optimum speech and optimum music conditions.

Initial Time Delay Gap and Ratio of Early-to-Reverberant Sound

These quantities have been analyzed primarily with respect to concert hall acoustics — although there is every indication that we should consider them important in large speech halls also.

In a large concert hall having an "ideal" reverberation time characteristic (a long one, say 2.0 seconds) there can be a tendency for the sound heard by a live audience to be *muddy* or lack sufficient clarity, particularly for classic (Mozart — Haydn) and contemporary (Stravinsky) music. Rather than shorten the reverberation time to obtain the added clarity, today's acoustical engineer would rather supplement the direct sound with additional sound energy that arrives at the listeners' ears soon enough to reinforce the direct sound, adding to clarity without destroying the richness of a long reverberation time. The goal is a "have your cake and eat it too" solution, which combines clarity and reverberation. Such a hall can have a large-hall liveness of sound with a small-hall intimacy. The time of arrival of these early reflections and their strength is important. If too late (after the initial sound) or too low in strength, they will be ineffective in aiding clarity: if they are too strong — and early enough — they will so dominate the sound heard by listeners that the liveness of the hall simply will not be heard, and the hall will have a reputation for *dryness* despite an adequately long reverberation time.

These early reflections may be compared to a pinch of pepper in a well-prepared soup. Just the right amount is needed. Too much and we'll taste only the pepper (too *much* clarity); not enough and the pepper may as well be absent (too *little* clarity); and the pepper should be added at the right time for best effect (the proper initial time delay gap).

The amount of energy in the early reflections, together with the energy of the direct sound, determines the ratio of early-to-reverberant sound, while the timing determines the "initial time delay gap". The ratio of early-to-reverberant sound is, for our purposes, defined as the ratio of sound energy received at a listener's position for the first 50 milliseconds during and after the receipt of the initial pulse of sound to the total sound energy received after the first 50 milliseconds.[3] Often, the inverse ratio — the ratio of rever-

3 We assume use of a short-duration (5 milliseconds or less) pulse or impulsive sound source to obtain measurements of the quantities here discussed. Such sound sources would be analogous to transient sound in speech and music — the attack of a musical instrument or consonants in speech — rather than the stead-state or vowel sounds. Proper hearing of transients is necessary for adequate clarity.

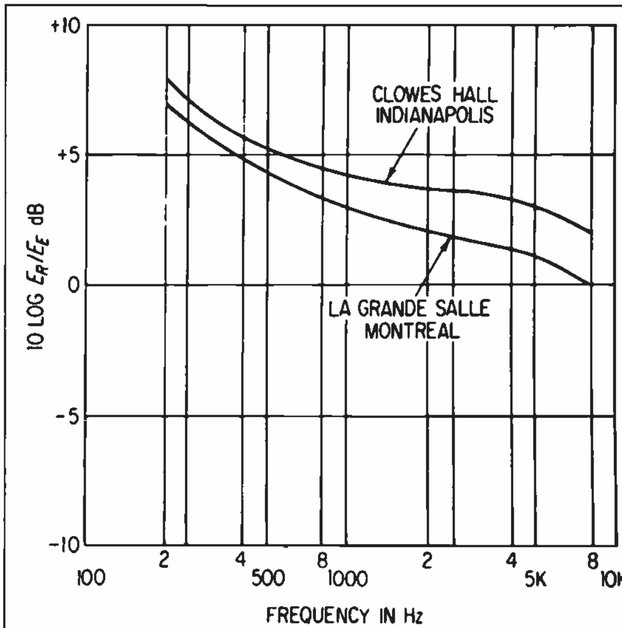

Figure 5. Ratio of reverberant sound energy to early sound energy of modern halls, expressed in dB (10 $\log_{10}$) as a function of frequency.

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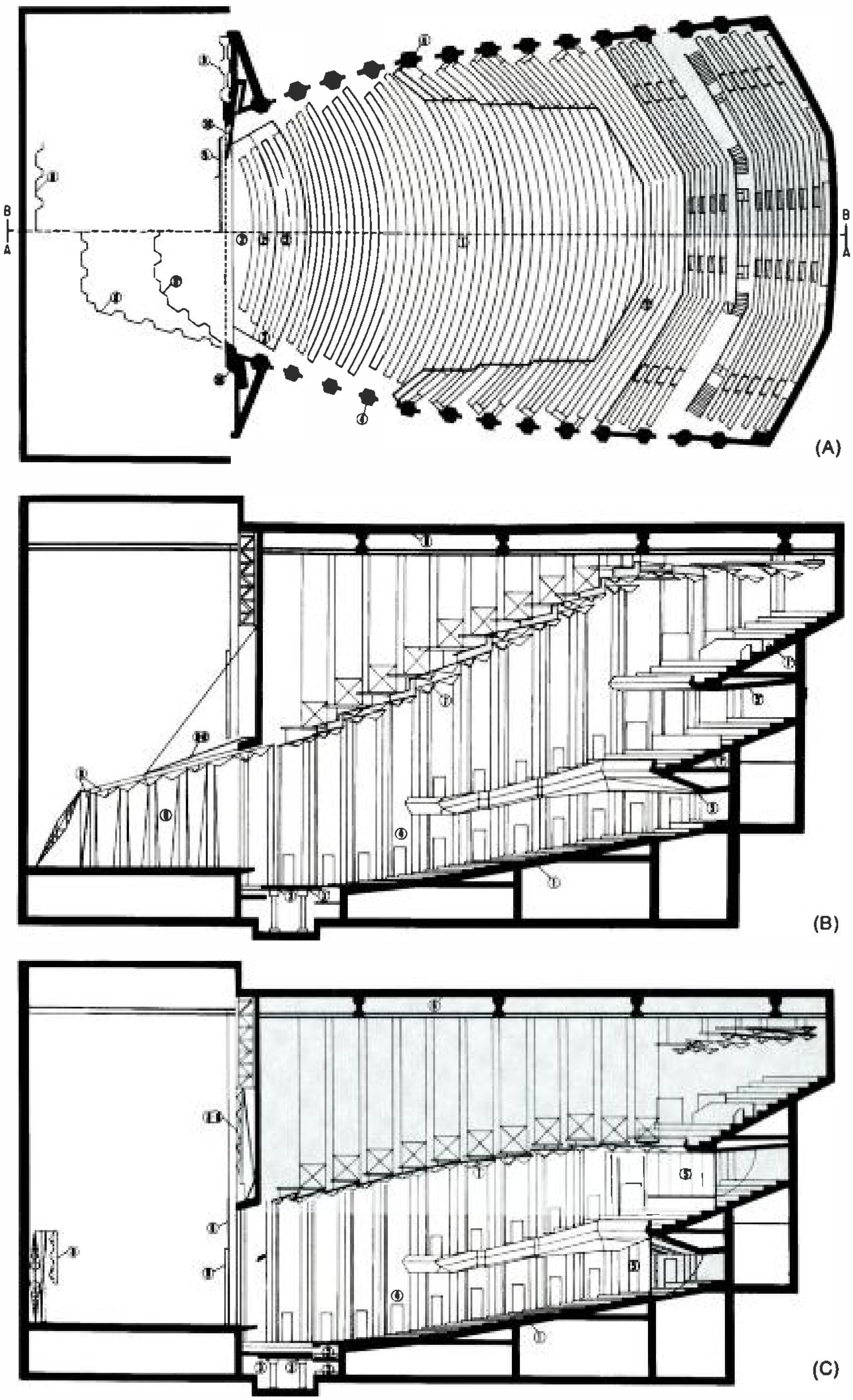

Figure 6. A 3000-seat Concert Hall-Opera House and 1700-seat Recital Hall-Theatre Combined in One Hall. (A)—Plan—left stage enclosure in position for 3000-seat concert hall, right, 1700-seat theatre, stage enclosure stored. (B)—Longitudinal section, stage enclosure in position for 3000-seat concert hall. (C)—Longitudinal section, stage enclosure stored for 1700-seat theatre.

1—Fixed Orchestra Seats 1′—Fixed Balcony Seats 2 and 2′—Movable Orchestra Seats 3 and 3′—Stage Apron Elevator 4—Fixed Auditorium Walls 5 and 5′—Sound-Transparent Movable Auditorium Walls 6—Fixed Auditorium Ceiling 7—Movable Auditorium Ceiling 8—Movable Stage Enclosure, Walls and Ceiling 8′—Movable Stage Enclosure Walls, Recital Position 9—Stage Teaser and Tormenter 10—Proscenium Pull-out Panels

Shaded areas are spaces closed off by movable ceiling and rear walls.

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berant-to-early sound energy is used. The ratios are usually expressed in decibels rather than fractions. FIGURE 5 shows the ratios for two modern halls as a function of frequency.

The initial time delay gap is the delay between receipt of the initial sound and the receipt of the first reflection. Beranek's study (see Ref. 4) has indicated 0 to 20 milliseconds as ideal for the initial time delay gap, and up to 30 milliseconds as good.

The proper early-to-reverberant sound ratio and the initial time delay gap were inherent in the design of the older, narrow, rectangular "shoe box" concert halls. The designs included broken-up or diffuse side walls, spaced sufficiently close together (and to the halls' center-lines) to provide the required early reflections. However, the newer, larger and sometimes more radically shaped wide halls seem to require supplementary sound-reflecting *clouds* (Tanglewood; Clowes Hall in Indianapolis; Jones Hall in Houston; La Grand Salle, Place des Arts, Montreal; DeDolan Hall, Rotterdam; for example) or one large sound reflector per hall in the form of a "lip" (Saratoga Springs; Chandler Pavillion, Los Angeles; Opera House, Seattle) to assure the right balance between clarity and liveness.

The recording engineer may not be overly concerned with these balances. Given a sufficiently live hall, he can increase clarity by closer microphoning or use of more directional microphones; or increase liveness by use of more omnidirectional microphones or greater distance from source to microphones.

The sound-system designer may broaden his understanding of room acoustics problems to have a better grasp of the combined room-sound system results. In special cases, where architectural means are impractical, an electronic sound reinforcement system may be called upon to simulate early reflections by amplified sound.

Stage Enclosures

Auditoriums used for opera and plays usually have generous fly-space over the stage and side-stage areas to permit movement of scenery and curtains. These voids can act as traps for sound energy during live music performances; the proscenium (stage opening) acts as a barrier between the musicians on the stage and the audience. A music studio or concert hall surrounds an orchestra with many hard, sound-reflecting surfaces. Movable stage enclosures accomplish the same results in multi-purpose auditoriums, they help in the "quick-change act" between theater and concert hall. When in place, the enclosure assures that the audience and performers are in the "same room".

To be fully efficient, such enclosures should usually be constructed of fairly heavy material, usually weighing 2 lbs./sq. ft. or more, to reflect low-frequency sound energy which can pass through thin, lightweight material. Half-inch or three-quarter-inch plywood is a popular material for stage enclosures, both custom-built for particular halls and available from stock from several manufacturers.

Good critical reviews have greeted concerts performed within relatively lightweight enclosures. These enclosures employ a process which has been described as *selective absorption* to achieve optimum frequency response and instrumental balance. Gaps are left in certain areas of the enclosure to absorb middle and high-frequency energy, balancing the absorption of bass energy by thinner than usual material.

The usual system for the erection of a movable stage enclosure is for the ceiling pieces to be suspended on the theatrical rigging system, with the walls in the form of interlocking self-supporting pieces or rolling towers. There is a trend, however, to mechanized stage enclosures that reduce dependence upon normal stage-hand labor for erection and striking while freeing the rigging system completely for its prime purpose of storing scenery. In the Jesse H. Jones Hall for the Performing Arts, Houston, Texas, the forward ceiling panels of the stage enclosure fold down as one unit from the stage house wall above the proscenium, the side-walls fold out from the stagehouse walls on each side of the stagehouse, and the rear wall and rear ceiling fold out from the lower up-stage (rear stagehouse) walls. All this is accomplished with the help of push-buttons and electric motors. Such a theater design can free the acoustical designer toward the use of adequately heavy materials, regardless of their weight. In this case, 12 gauge steel, backed with damping material and faced with thin wood veneer was employed. The surface weight of the combined structure is close to 7 lb./sq. ft., assuring good sound reflection down to 25 Hz.

The improvement of an existing mutlipurpose theater-concert hall by provision of a properly designed stage enclosure, whether wood, plastic, or damped metal, is a familiar event today. Often the improvement in hearing conditions for the live audience is matched by improvements in liveness and balance in the sound as picked up by microphones; however, more often than not complete re-engineering of the microphone pickup arrangements are required if best results are to be obtained.

This article discusses the room acoustics aspects of studio, auditorium, and concert hall design. Future articles will touch on sound systems design, noise control, and sound isolation, all important parts of the over-all architectural acoustics picture.

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The Commercial Sound

by Peter Burkowitz

Contributed and Translated by Stephen F. Temmer

A paper delivered before the 7th Convention of Studio Operating Engineers held at Cologne, Germany October 1966 (Tonmeistertagung) under the sponsorship of the Detmold Academy and the West German Broadcasting System (WDR) Cologne, W. Germany

WHEN I was asked to give a talk concerning the problems of the commercial world of a record company, I found it not too easy to select the proper theme. After some thought, I came to the conclusion that it would be of considerable interest if I investigated the problems which occur at the boundary points between the engineers and their non-technical partners—in

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other words, the areas of the business people, the sales people, the marketing people, the producers and the artists, the stores and the customers—and to see where our mutual problems overlap or perhaps even collide.

As far as the exact technical problems are concerned I only want to make the brief determination that they are the professional life to the engineer, spooky and too expensive to the business and marketing man, a source of annoyance to the producer, suspicious to the artist, and, on the whole, mysterious to the distributors, dealers and clients.

But I did want to speak about an area which borders on the activities of the studio engineer. In this regard, I considered music production as the most immediate area from the standpoint of the engineer: What does one have to ask of the performance and what can one expect of the engineering? The title "Commercial Sound," which might sound somewhat profane to strictly professional ears, was not chosen to insure a well attended lecture but rather because it is always those people who cry loudest for "commercial sound" who are unable to see clearly the boundary between musical content and technical packaging. As Dr. Steinhausen[1] once said so poignantly, "it is just those people who during and even after the recording session expect from engineering a replacement for what *they* were unable to accomplish."

The following descriptions should not be taken as a polemic, but rather as an attempt to bring to this division between artistic offering, manufacturing, and the activities of the artistic-technical personnel which encompasses both activities, some sort of guide-posts—perhaps even as a small bit of education for future producers among you.

First of all, let us look at the situation as it always should be. To those who might think me naive, I would like to explain that this example is obviously somewhat idealized. It starts with a producer who only concludes contracts which he can live up to, with artists who represent, in actuality and not only reputedly, significant talent in the role they are to play. It follows that these artists are given the opportunity to fully know their part on the day of the recording session, and that the producer, director and perhaps even the studio engineer had previously coordinated, controlled and corrected their individual roles in this session. The orchestra too will have been rehearsed diligently in such a well functioning production, especially if its role in the recording is a sizeable one.

For realizing the best results from this performance, one will select an excellent hall with the best acoustical properties since the resulting sound is next in order of importance after the value of the composition and its interpretation.

One will then conscientiously choose the physical setup, and, hopefully, make clear to the fully assembled performers, in calm terms, the importance of the impending unique performance. This part of the task is, incidentally, the responsibility of the producer. In such a production, in which nothing whatever goes awry, the performers long before have been precisely instructed as to what is expected of them in front of the microphone. The producer will not attempt to gel the setup during the session, possibly even against the desires of the artists. In the control room there will be no senseless debates about hoarseness, sibilants or presence of the tympani sound, and in the end there will be a minimum of required editing.

I don't want to go into the many equally important details involved. This record will, no doubt, become a major *happening;* it will be well reviewed by the press and will become a world-wide success. But above that, a recording *must* be made in this way today if it is to enjoy *any* success.

What happens after this point is a matter of precise engineering. The fact that engineering will know the requirements all the way to the mass-produced disks and tapes is a matter for which we rely on the professionalism of the engineering community. In the engineering profession certainly, nobody will measure output level with a manometer, or tape speed with a light meter. This might sound banal to you, but parallel things occur in the aforementioned professional disciplines now and then. But I do not wish to give the impression that the complex requirements of recording production could be served using a simple set of devices. In the quest for more cooperation between the various segments contributing to such a project, I want to point out the vast gap between the methods of commercial technique and engineering technology, which would not exist if the artistic side of the effort would utilize the same systematic methods that are used in engineering. I am convinced that even outside the world of engineering it is possible—and logical—to eliminate accidents and failures, for here, as there, all it costs is money.

To make all of this more graphic, allow me to construct a negative example. First of all, the producer sees himself in a bad situation since most repertory ideas are considered white elephants to begin with. He therefore tries to switch to a strategy of "special interpretation" of a work which already has been recorded in abundance. Such special interpretation is almost always restricted to the world of *stars* and first-rate orchestras. While searching for such stars, the producer soon discovers that very few suitable artists can be found in his own stable. Aside from that, he probably will have considerable contractual problems with Miss X and Mr. Y. So he arrives at a recording date with such a compromise cast that everyone feels a miracle would have to happen to make this recording anything more than a mediocre effort.

In the spirit of this example, there will doubtlessly be available at the designated time and place only one suitable hall or studio, whose sole claim to suitability will be that it has enough square footage to accommodate the entire cast. Besides all this, we must assume that there have not been sufficient rehearsals, that the soloists are not altogether familiar with their parts, and that the conductor is scheduled to conduct a concert only three hours hence. Aside from the fact that this should be sufficient to cancel the whole thing, the producer now commits another serious error: he thinks the engineer will be able to evolve a brilliant recording from these haphazardly assembled fragments, using his magic conglomeration of machines.

At this point, though, I would like to say that nowadays even the production side of the fence is relearning an old truth: Artistically demanding and commercially viable recordings cannot be obtained by indiscriminate editing or "cooking with acoustic artificial flavoring." Even if a valuable work is praised now and then which has been recorded by overwhelmingly excellent artists but with inadequate sound realization, this is no proof against that "old truth." The most skilled jeweler would not, in the long run, stay in business if he offered his valuable gems in tin mountings.

The next mistake is committed if, after the recording gets under way, it turns out that the conductor, the producer, and the artists have entirely different understandings of the

[1]Director of Engineering; Deutsche Grammophon Gesellschaft, Hanover, W. Germany

piece, and if the producer now tries to convert these people to his point of view.

The next error is committed if the recording dates are so arranged as to make it necessary to record small segments without any logical sequence. The next, when the call goes out from this now surely nervous group of performers for the "artistry" of the engineer which always salvages everything, and the last, when this mish-mash is actually offered for sale.

Of course I have painted a rather bleak picture, for naturally such a coincidence of failures will never occur in major productions. On the other hand, it is useful to remember that a *producers' manual* does not exist. Many of you could find yourself in a position some day to produce or direct such sessions, and what better way to learn than from both good and bad examples.

But let's get back to the moment where the man at the console – and this predominantly for pop music – is asked that he make the sound more *commercial.* What is meant by that? Well, the performers would like their voices and instruments to sound so beautiful that the buying public will literally tear the recordings out of the dealers' hands. They mean that instruments, basically, are instruments and voices, basically, are voices. Therefore, they assume it should be possible to record every instrument and every voice with bell-like clarity, since they have already heard instruments and voices for which this has been done perfectly. It follows, according to the artist, that it can be only a matter of engineering if Miss X or Mr. Y doesn't come out that way. Perhaps the compressor is not aligned properly or the microphone is too high? Couldn't it be that 3 dB more highs have to be set? And before you know it, the control room has become a laboratory. The sweating engineers drag in ever more gadgets and black boxes. Engineering is king, while artists and producers watch with approving glances. And somehow after all that, things *do* sound better. (In any case, everyone convinces everyone else that it is so, and after hours of monkeying around nobody has any idea anymore how the whole thing sounded to begin with.)

At the end, both the tired-out producer and artists will pat the engineer on the back and will not skimp with praise. Futhermore, they will tell him that he, and only he, is capable of handling their recording the next time.

If this engineer doesn't wake up the second time around to what is being played here, then he is well on his way to selling his "mixing soul" to the devil and once again is helping a producer and an artist to live a technically subsidized sham life.

The end then looks something like this: the artists, rather than strain at all, record heaps of performance pieces for the engineer in a slovenly manner, relying on his technical eagerness to magically produce a recording for them. Worst of all, these producers and artists find nothing wrong with this, but on the contrary, will praise this engineer as the true perfection of his profession.

Here, I do not exaggerate. I have actually experienced such cases, and the people involved didn't even notice it. Naturally the results of such a method are no better (and more importantly) no more saleable than solid artistic workmanship. As soon as this fact has been realized, the eager engineer in our example will have to look around for new technical tricks if he intends to continue to enjoy the praises of the producer and artist whom he himself has brought up this way. This is a vicious circle concerning which every aspiring young engineer should be informed *before* he undertakes this profession. He can prevent it from ever coming to this point if he convinces the artists in a firm but friendly manner that the steps necessary to improve their artistry have to be taken by them alone. The artists, on the other hand, won't ever think of taking over his technical duties.

The young engineer should be aware that equalizers, limiters, compressors and the entire instrument pack available to him are intended to effect *quantitative* corrections – to correct for technical insufficiencies of the studio, the recording medium, the disk limitations – not, however, to improve upon the quality of the sound source! The sound source can not be improved, since that which needs correcting hardly ever lies in the technically controllable ranges. Should the technical-equipment arsenal nevertheless be brought to bear on problems which are actually created by lacks in artistry, we then get the well known sound conglomeration characterized by flattened dynamic range, shrill presence, and pumping reverberation which is flaunted by many professionals, especially in the field of pop music, as being the prerequisite for market success.

In the field of serious music, the problems are a bit clearer, although we find some tendencies even there to apply exaggerated technical tricks. Many modern-day recordings with metallicly sharp treble and unnatural proximity of sound sources to microphones testify to that fact.

What the crux of the problem is, was most dramatically demonstrated to me in the presence of several of my colleagues some years ago. Two orchestras were to be recorded in the same hall using the identical microphone setup, one after the other. The musicians of the first orchestra frequently asked after playbacks whether the strings couldn't be set louder and more brilliant and the basses less muddy (they believed these to be technical inadequacies). The sound nuances of the second orchestra were perfect right from the start. The difference lay solely in better instruments and playing technique.

And so we have arrived at the nucleus of the theme: The *commercial sound;* – that fascinating sounding, and therefore more saleable realization of a composition – is nothing else but the well-aimed coincidence of optimal performance attributes *in front* of the microphone. It is assumed, of course, that perfect engineering facilities are involved, but no more than that! The secret of success (and with it the business reward) in recorded music of all types, lies then as now in the culmination of talent, phantasy, temperament, and perfection in front of the microphone, and simple technical precision behind it.

In the future, the recording business must introduce to recording production, as well as to the research into and catering to the market, the same logical methodology which dictates the world of science, law, and administration. In an attempt to make a method of success, record production will not slacken its efforts, but, quite the contrary, will try to replace empirical methods with systematic ones, even if this means giving up some degree of enjoyable freedom of action.

As technically trained and musically interested professionals, we can be of significant help to our colleagues in the A & R department in this process. In the final effect, we can help in reducing costs and increasing the chances for success. In short, we can improve the product of our common task by means of technical and business systematics. It is in this field of influence, and not solely by the skillful operation of consoles, that the important, valuable, and superior professional task of a studio engineer lies.

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Audio Facilities—Planning, Construction and Servicing

Following are the highlights of a meeting held by the New York Section of the AES this past September.

The panelists were Richard W. Burden, president of Burden Associates, consulting engineers; Henry Krochmal, facilities engineer with NBC; Joseph Giovanelli, president of Audio Tech Laboratories, specialist in disc mastering and long a columnist for Audio Magazine; and Abe Kobrin, with Harvey Radio for twenty years and the foremost technician in the New York area on Ampex professional products. Irving Joel of Capitol Records was the chairman and moderator. The transcript has been edited somewhat to improve clarity.

Before the question and answer session that comprised the bulk of the session, each panelist made an opening statement.

KROCHMAL: My role is in the facilities engineering department as a designer and engineer. Whenever there is a need for a new piece of equipment or a system – audio or video system for studios or camera chains, or any system that requires components, we are given the job of putting something down on paper that someone can build. When the job is first assigned to us we get an idea from the operating people as to what the system should contain. This is logical; after all, they have to operate it as comfortably as possible, and want it trouble free. It's best to go to these people, of course, but it must be remembered that you just cannot add everyone's desire for everything or you'll end up with a console that won't even fit in the studio. So you weave through all the information, colate it, and make up what you think will fit the bill for most of these people.

The final design must be not only comfortable—that is, easy to operate – but also it must present few problems as to maintenance (since down-time could run into thousands of dollars).

The discussion panel. Left to right seated—Abe Kobrin, Joe Giovanelli, chairman Irv Joel, Henry Krochmal, and Dick Burden. Art Gruber, standing, is introducing the panel.

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Once a rough design is down on paper we build a mock-up, usually of cardboard. This gives the people who will ultimately use it an idea of what it looks like; how it will fit. If there are no further changes to be made we go right ahead and engineer the system. Most of the time the job will be turned over to a construction firm. On occasion when the job is to be built right here (at NBC), we hire people to do the constructing.

Burden: When a construction company such as we are comes in we take Henry's basic idea, sit down with him and his paperwork, look at his mock-up, and go back and read over his ideas again. Then we come up with a list of questions:

Why do you do this? Why do you want that? In this way a design engineer from the construction company's side of the fence can get a pretty good idea of what someone like Henry wants this console to do. Then we can start without preconceived notions as to how this unit is to operate. And when we get to the final design we may find that something must be rearranged because this is the only way we can fit the necessary components together.

Today's industry is going more and more toward modular type construction. There often isn't time during the day-to-day operation to look through a console, trace a wire, or run into an amplifier to replace a capacitor. We have to be able to check quickly, pull the faulty amplifier out of the circuit, and replace it with something—fast. So the item must be accessible as well as replaceable. If we are going to make a modular arrangement, it will not do if maintenance must go down beneath the operator, through a trapdoor and down a set of stairs to get to something.

So the system must be set up with quick accessibility in mind. This will sometimes require moving a few things around on the panel. All this must be done with some thought to the operator himself so that he doesn't have to be a foot-and-a-half tall with an arm reach of seven feet to be able to operate this gadget. We want to make those sections that we have set up for preventive maintenance particularly accessible. The logic behind this dictates that hard to reach areas will be skipped during routine maintenance – and the next thing you know you have a failure.

All of these factors must be considered before actually building a piece of equipment.

Kobrin: After Henry and Dick have completed their jobs we have the ever present problem of maintenance. It hardly seems necessary to stress the importance of proper maintenance. So allow me to give you the difference between proper and improper maintenance.

We had a man bring in a recorder with the complaint that it didn't work. So we opened it. We found that he had read his instruction book carefully. It said that he should put three drops of oil on the capstan bearing – so he did. He liked this so much that he put three more drops in. This looked very good so he put three drops on the idler bearing. And this looked very good so he put three drops more there.

"Well," he said, "this looks terrific." So he put three drops on the heads because the tape runs over them and he put about four or five drops on the screws and five or six drops on the meter because it wasn't glossy enough. Well, by the time it came to us it was one massive glob of oil. Now the clincher is that this wasn't bad enough, but the darn thing smelled of Mazola!

This is what I consider improper maintenance. If he had really followed his instructions properly and used the correct equipment, this machine would have run properly.

Giovanelli: If we have done our routine maintenance correctly, the odds are that any emergency is a rather minor failure. With proper maintenance being done, you are not going to have a general catastrophic failure somewhere in a feedback loop. It is a diode that has popped in the power supply, an electrolytic which has shorted; now these things do not call for esoteric tracing methods. We can look and see where the smoke is coming from. Most of the time we can fix something quickly using nothing more than a pair of headphones and a couple of alligator clips. Even without any equipent we can usually do it. You don't need an oscilloscope to find that there is no B+; you don't even need a volt meter much of the time.

All I'm trying to say is look for the problem easily. Don't get so involved in theory and analytic work that you lose sight of the fact that you didn't turn the switch on or put in the plug. It happens.

This concluded the formal opening statements. What follows are selected audience questions posed to the panel.

Question: In the wiring of facilities there are two methods in common use. One is known as the brute-force ground method, where you use non-insulated shield wire bundling them together. The other is the method using insulated wire bundled together with a grounding at one common point.

I'd like to know from the panel which they prefer – and why.

Krochmal: It has been my practice and my feeling to use shielded wires but no jacket at all. (I assume the type of wire referred to in the question is shielded wire with no jacket on it.) It makes wiring much easier. I would say that virtually all of the time it is entirely satisfactory. The one thing we find that we must watch carefully in our installations is where we terminate those wires.

We bundle wires depending on what level audio we expect in those wires. For example, keep all microphone-level conductors in a single bundle, but keep that bundle at least one-half to one-inch away from a bundle that might be carrying line level. At NBC we use the three shielded bundle forms: mic level (around −60), fader level (around −22), and line level (around +8). We simply run them parallel to each other but not bundled together. When they terminate at a block we make sure that our lugs in the block are assigned in such a fashion that there is never mic level on a lug adjacent to one carrying line level. We will, in fact, often use separate blocks for the purpose. With this system we don't run into trouble at NBC. However, I do see where r.f. pick-up could be a problem when you are in high fields. I think that Dick Burden probably has a different point of view on this.

Burden: It's r.f. that is the problem, particularly at the a.m. stations where you get a length of wire that all of a sudden is the right length to become a half-wave antenna and pick up r.f. Bundling wires together, though it certainly produces a good ground, can put you in this position. Grounds are an odd thing; it depends so much on how and where your cables are connected.

What we do is usually leave the shields to a certain point along the line. Then we decide where they are going to tie, making sure that they are only tied on one end. Once you start to bundle, if you have tied a cable by mistake in the wrong place, you've got a real problem on your hand. My preference is definitely to insulate, particularly if there is an r.f. field.

Joel: I think it has to be remembered that in grounding practices you must realize that we are dealing with two different types of technology. There is *broadcast procedure*

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where everything is balance that comes and goes. And there is also *unbalanced circuitry* that is inherent in transistor equipment. Grounding procedures have changed considerably since the advent of the transistor and its type of circuitry. In any case, you cannot intermix the two systems. You must decide which way you are going to go – and stay that way.

Question: It should be added that most of what has been said refers primarily to balanced circuitry where two wires are shielded to a conductor. But what are the general ideas today when using single-ended circuits where some of the time the shield is carrying part of the actual signal or the ground side of the signal circuits.

Giovanelli: You wouldn't want your shield to carry the ground signal. You'd rather have it come back almost as if it was balanced, where you have a second conductor to bring the signal back and then tie the shield at one end somewhere – assuming you could, of course.

Burden: We treat it just as though it was a balanced line.

Joel: I was going to say earlier, travel your audio as *high* and *low* audio. Don't confuse it with B+ or B–.

Burden: A good example of this is that you might run into a bunch of ground loops at the actual installation site even though none were present when the system was setup in the shop. If you have used this type of grounding procedure with two wires hanging, you can use an isolation transformer to get rid of the ground loops. Once you have used the shields for grounds you are finished as far as putting a transformer in.

Question: It has been mentioned that there is signal-conductor shielded and two-conductor shielded. Now in referring to microphone lines of 50-, 150-, and 250-ohms input through a transformer, there is another method using three wires. This uses all three run through with the shield connected at one end only. Now do any of you have an opinion of this as compared to two lines?

Krochmal: Our practice, with mic circuits particularly, is to ground in only one spot. That would be the point close to the terminal or amplifier equipment.

As an example, we have just completed a job where we had long mic lines on the order of six-hundred yards. (It was a golf match.) We used no grounds at all except right at the receptacle panel where the mic cables were plugged into the truck. This raised no problems. If we had connected at the far side as well we would have had ground loops.

Question: With a run like that wouldn't you want to have a preamplifier right at the microphone, thus overcoming the problems by boosting the level right there?

Krochmal: Yes, that would be very fine, except that we had thirty-nine mics.

Joel: There is an NAB standard that spells this out very clearly.

Question: Quite a bit has been said now about shielding. But one worry that remains is: where do we get our ground terminal system; do we go out to a water pipe or some such in a tall city building?

Burden: The water pipe is not necessarily a good answer. It might be better than nothing but sometimes it makes a fine antenna picking up your competitor down the street. In some cases, a heavy cable is simply brought up to the level you need it. It can be even a 00 Greenleaf, jacketed. Then bring that up inside a conduit. Ground the conduit and the wire at the bottom end. The conduit becomes a shield eliminating r.f. pickup by the ground wire (to some degree).

It must be added that we are dealing with a considerable problem. There are no simple answers that can be given.

Question: Where do you finally ground to when you've taken that conduit down? There are different levels even of earth grounds?

Burden: In New York this is a very real problem since we are on rock. You simply have to find the best ground you can get. In a broadcast station you may want to ground right to earth through the ground plane for your antennas. Drop a good piece of copper and braize right to it.

Question: What do you do on remote locations where you simply must plug in wherever you are?

Joel: Bell System practices specify on remote location that multiple grounds be driven and the audio ground then be attached to this system. I've been out on remotes where we had to drive as many as twenty stakes before we got where we were going.

Remember that there are two basic theories to keep in mind:

Ground only to one place and ground everything to it. Or, ground everything to everything.

On remote locations the usual practice is to pick your location and ground everything to it. Only if you are in a small complex might you consider the ground-everything-to-everything practice.

Question: I'd like to expand on the question of wiring a bit more. In the past it has been the practice to use an audio system starting with a microphone, bringing it up to a patch panel, bringing it back down again to the mixer, bringing the mixer to the patch panel, back down to the amplifier – this could go on forever. Do you feel that this is still the way to go?

Krochmal: At NBC we still do this, mainly for flexibility purposes. We do not do what many recording studios do; there isn't a filter for each mic channel. We have four patchable filters in one of our consoles and we make it patchable to any one of forty-two mics. Four is the maximum we will ever need, but we don't know where we will need them.

Giovanelli: I would like to add that I prefer the system of coming out to a patch panel so that I can plug something in if I want to listen to it; or feed a signal into it just from a maintenance standpoint.

Joel: Let me just air both sides by saying that if you could conceive of a module that would do everything you could ever ask of it by switching, or by duplicating whatever equipment was necessary, then it would be feasible to remove some of the patching facility. Some of this is being done successfully because of such things as operational amplifiers. Where it was formerly quite expensive to build something such as an equalizer, now professional amplifiers costing extremely little – under $10 – are capable of doing some of the equalization. It then becomes feasible to put these in multiple into the console's modules. When you do this, you can eliminate some of the jacking.

Everytime you remove a jack you remove a point source of noise because in most systems these jacks are normals; thus they are a potential source of noise. And with unbalanced circuitry in the module, you are able to live with fewer transformers in the chain (looking at it from front-to-back). This is certainly a design consideration.

Burden: This is going back to the idea of planning before anything is done. Ask yourself or someone else if you really need all the things you are asking for. Talk to someone on the outside and you may get a fresh view. I've found that in many cases people ask for a lot more than they need. They *over specify*. After discussion we may find that 10-15 per cent less than was thought is actually needed.

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New Products and Services

Color Brochure on Vtrs

● An eight-page color brochure describing features, specifications, and operation of the VR-2000B high-band color vtr is available. The recorder is designed for use by tv stations and production houses for high-quality color recording and sophisticated teleproduction. It results in multiple generation copies equal in quality to a master.
Mfgr: Ampex Corporation
Price: no charge
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Production Mood Music

● The 1968 Major Records catalog offers thousands of classifications minutely detailed as to timing, content, and availability of musical and sonic selections for use in all areas of sound production. Music is available in three forms: lp record, full-track tape, and transfer to magnetic sound stripe of 16 mm or 35 mm film ready for a mix.
Mfgr: Thomas J. Valentino, Inc.
Price: no charge
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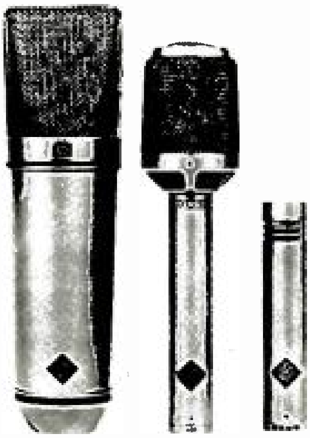
FET Microphones



● The FET-80 series of mics includes silicon solid-state versions of several Neumann types. The U-87 is the new counterpart of the U-67. The KM-84 is the FET-80 version of the U-64. Accompanying these units is the KM-86, the new version of the KM-66. There is also the new KM-85, the updated KM-84. It has low-frequency roll-off for close microphone and sound reinforcement use. Of these microphones, familiar features have been retained. The U-87 is three-pattern switchable with additional switches for overload protection and proximity correction. The Neumann KM-86 is a miniature three-pattern, side-addressed mic. Two U-64-type capsules are mounted back-to-back. This series' mics are available in systems to be powered from the N-452 dual mic a.c. powered unit or the BS-45 battery supply accessory. No r.f. circuits are used; element polarizing voltages are obtained directly from the supply voltage.
Mfgr: Gotham Audio (dist)
Price: Will be supplied
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Mobile PA Amplifier

● 40 watts of power distinguish this five-transistor, one diode unit. Three separate, illuminated fiber-optic controls have individual on/off and gain functions with mixer/fader circuits permitting separate or simultaneous playing of external equipment and a paging system. Three inputs: one low-impedance mic, one high-impedance phono (ceramic), and one aux. Uses 12-14 V d.c. supply source. Includes mobile mounting bracket and d.c. cable. This is catalog model 44-0140WX.
Mfgr: Lafayette Radio Elect.
Price: $69.95
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Sound Mixing Systems

● The AM4A systems feature blocks of pre-wired, plug-in modules. The low silhouette provides the operator with a convenient sloping panel. Overall height is 7 inches and depth is 27 inches. Frequency response is ± 1 dB, 20-20,000 Hz at an output level of +8 dBm at 1 kHz (without l.f. or h.f. equalization). An equivalent noise level of −122 dB can be maintained with a system gain of 60 dB. The design center is a source impedance of 250Ω. Output load impedance is 600Ω at a standard output level of +4 or +8 dBm. All systems meet ASA and NAB requirements.
Mfgr: Langevin
Price: Dependent on system
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Monitor Amplifier

● The first in a series of new amplifiers is this Model 610. Of all silicon-solid-state design, 10 watts of r.m.s. power is available with total short-circuit and overload protection. Features include compact design and low distortion. There is an integrated power supply. A barrier strip on the rear provides all input and output connections. Specifications include: 10 watts into an 8-Ω load, 6.5 watts into 16Ω, output impedance is 0.5Ω, damping factor is a minimum of 16; input impedance is 10k bridging; and the required input level for full output is −10 dBm. Frequency response is ± 1 dB, 20-20,000 Hz; t.h.d. is less than 0.4 per cent at full power; noise is −86 dB below rated output.
Mfgr: Fairchild Recording
Price: $138.00
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New Recorder Line

● The model 500A professional recorder is designed for broadcast studios, program automation, and educational institutes. Meeting all NAB requirements, it offers two-directional record-and-reproduce capability. All operation can be remote controlled. Separate high/low torque switches are provided for each reel drive motor. The capstan motor is two-speed and is available with any two adjacent speeds from 1⅞ to 15 in/sec. Mono and stereo options for full-track, half-track, two-track, and four-track operation are available. There is also a single-speed logger model with 5/16, 15/32, or 15/16 in/sec. speeds.
Mfgr: Metrotech, Inc.
Price: $1695.00 (two-track stereo): Automatic Reverse Units are $1895.00.
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Uher Promotion

● Not strictly in the realm of a new product—but broadcasters and other users of remote recording equipment will want to know that a special promotion exists on the Uher 4000 Report-L battery-powered professional recorder. This unit, normally sold by dealers at $440 is being specially sold *only through June 15th* for $100 less ($340).
Mfgr: Martel Electronics
Price $340 till June 15.
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People, Places, Happenings

● At **Perma-Power, James F. White** has been appointed vice president, **Ampli-Vox Products,** according to **Richard Goldstein,** Perma-Power president. As vice president, a newly created post, Mr. White will direct a force of some fifty sales representatives and coordinate the sales activities of approximately 4,500 salesmen affiliated with 600 authorized Ampli-Vox outlets. Mr. White comes to Ampli-Vox, manufacturers of sound systems, from **Rheem Manufacturing.**

● The annual convention of the National Association of Broadcasters will be held on March 31st through April 3rd at the Conrad Hilton Hotel in Chicago. There will be much of interest to **db** readers. On the first day — designated FM Day — several new horizons will be discussed. Among these will be the growing use of fm car radios, operations in both small and larger markets by newer stations, on-channel boosters, and dual polarization. Special reports will be delivered by Harold I. Tanner, WLDM, Detroit, chairman of the NAB FM Radio Committee and Charles M. Stone, NAB v.p. for radio.

The last day will feature a luncheon for the Broadcast Engineering Conference part of the convention. At this luncheon Howard Chinn, director of general engineering for CBS Television, will receive the NAB Engineering Achievement Award for his leadership in helping to develop NAB's new Standard Loudness Reference Recording. This standard has provided broadcasters with an additional means of maintaining proper audio levels between commercial announcements and programs.

There will be more awards and special features between these two ends of the show. Most of these will go to people involved in programming rather than engineering aspects of broadcasting.

The main attraction for many is the broadcast equipment manufacturers' exhibits. This year, 54,000 square feet of space, 4000 square feet more than last year, will house the 126 manufacturers signed up as of late February.

All exhibit space is at the Conrad Hilton Hotel in the Continental Room, the East, West, and North exhibit halls, plus two new areas — the Normandy Lounge and the Writing Room.

The exhibition rooms will be open the entire time of the conference. At press time the only hours that were firm were the opening at noon on Sunday the 31st and the final closing at about 5 p.m. on Wednesday the 3rd.

Technical meetings will be held throughout the conference hours at the Pick Congress Hotel, a few blocks from the Conrad Hilton.

db expects to publish the most interesting papers given at these sessions in coming issues.

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Used recording lathe—must be in perfect condition and reasonable. Please send photo if possible. Box B3 db Magazine, 980 Old Country Rd. Plainview, NY 11803

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● It's a busy April for audio pros. Sandwiched between the NAB Convention at the beginning of the month and the AES at the end, is the **Midwest Acoustics Conference** to be held in Evanston (near Chicago), Illinois. This is a one-day affair to be held on April 11th at Northwestern University.

TECHNICAL PROGRAM

1:30	J.A.D. Cooper	Introduction
1:40	P. Dallos	The Northwestern Univ. Acoustics Program
1:55	W.O. Olsen	Performance Characteristic of Hearing Aids
2:16	J.E. Jacobs	Ultrasound Image Visualization Techniques
2:37	R. Parmelee	Earthquake Engineering
2:58	E.R. Hermann	Environmental Noise, Hearing Acuity and Acceptance Criteria
3:20-3:45		COFFEE BREAK
3:45	K. Reimann	Piezo Electric Film
4:06	Z.G. Scheony	Binaural Hearing and Masking Level Differences
4:27	L.H. Pinto	Design of an Acoustic Impedance Bridge
4:48	S. Ramaswamy	Helicon-Phonon Interaction
5:09-5:30	K. Horikoshi	Heart Sound Analysis

The morning hours will be devoted to an open-house tour of the University Technicological Institute. In the evening there will be a dinner at the nearby Orrington Hotel. The featured speaker of the evening session will be Dr. Harry F. Olson on *Trends in Sound Reproduction Research*.

The afternoon and evening sessions are free of charge. The dinner is $5.95. For reservations call or write to John T. Williams, Hallicrafters, 600-08 South Hicks Road, Rolling Meadows, Ill. The phone is (312) 259-9600, ext. 303.

● **Visual Electronics Corp.** held a meeting of all its regional managers recently at the company's new vtr manufacturing plant at Sunnyvale, California. Highlight of the two-day conference was a preview of the new Visual line of vtr units. Included is a successfully tested model, the VM-90 slow-motion color disc-playback unit. In addition to manufacturing facilities, the new plant also has a vtr training school for the training of technicians in the installation and servicing of the expanded vtr line.

● **George Alexandrovich,** vice president and chief engineer of **Fairchild Recording Equipment Corp** has assumed the general managership of that company. During the eleven years he has been with Fairchild Recording. he has been responsible for the development of an appreciable number of new audio devices that are now a part of the company's standard line. His investigations into the use of light-cell technology in audio control led to the development of two complete input lines (one a module approach. the other a remote-controllable plug-in audio card series) that are the foundation of the company's present product structure. Under his direction the company is also actively engaged in the custom-console field. Mr. Alexandrovich is a member of **db**'s Editorial Board of Review and a monthly contributor.

● April 29th to May 2, 1968 are the dates of the **34th Convention of the Audio Engineering Society.** The place is the Hollywood Roosevelt Hotel. Los Angeles, California. It promises to be the best yet with many papers of wide interest scheduled. Don Davis of Altec-Lansing is chairman. The scheduled technical sessions and their chairman follow.

Acoustics and Hearing – Art Soffel, LTV Research Center.
Amplifiers, General–James L. Noble, Altec-Lansing.
Amplifiers f.e.t. – James F. Kane, Motorola, Semiconductor.
Audio Applications–Keith O. Johnson, Gauss Electrophysics.
Instrumentation–Allen E. Byers, Waveforms, Inc.
Music and Speech – M. V. Mathews, Bell Telephone Labs.
Recording – Charles Pruzansky, RCA Victor Record Div.
Recording and Broadcasting Facilities–William C. Dilley, Spectra Sonics.
Sound Reinforcement–Rolf J. Hertenstin, DuKane Corp.
Tape Cartridge Systems–Pell Kruttschnitt, Capitol Records.
Transducers–George L. Augspurger, J. B. Lansing.

The exhibition rooms will be crowded with 34 exhibitors (as of press time) including **db Magazine.** Next month we will have a visual guide to the exhibits as well as the complete program for the four days.

● **Edward J. Goodman,** president of **RPL (Recorded Publications Laboratories)** of Camden, New Jersey has announced the appointment of **Ernest W. Merker** as vice president, engineering and operations. Mr. Merker will have complete responsibility for the technical and aesthetic quality of original sound recordings, mass tape duplications, sound tracks for motion pictures and slide films, and specialized creative sound packages. As RPL's chief engineer for the past thirteen years, Mr. Merker developed, designed, and supervised the installation of studios and equipment for commercial, industrial, and educational sound applications. In his new post he will continue to supervise both the equipment and production aspects of existing audio-visual sound media. He will also explore and design new equipment and methods.

● New officers have been named for the **Institute of High Fidelity. John Koss,** formerly v.p. of the Institute is now president. Mr. Koss is president of **Koss Electronics Inc.** of Milwaukee, Wisconsin. **James J. Parks of Fisher Radio Corp.** is the new Institute v.p. Treasurer is **Walter Stanton of Pickering and Co,** former Institute president. Elected Directors include **E. L. Childs (Elpa Marketing), William Glaser (H. H. Scott Inc.), Walter Goodman (Harman-Kardon), Edward S. Miller (Sherwood Electronic Labs.), Harold Schulman (United Audio), William Thomas (JBL)** and **Mr. Stanton** and **Mr. Parks.**

Robert A. Strome, eastern sales manager of Ampex Professional Audio Products, died in New York City on February 22 at the age of 37.

We at **db** considered him a personal friend. Bob was widely known in the audio world. having previously been associated with the Fairchild Recording Equipment Corp. and the Marantz Company.

He enjoyed the respect and admiration of all for his wide knowledge and far-ranging interests in many areas of audio. One of the best-liked men in the field. Bob's lively company and interesting conversation will be sorely missed by our fraternity.

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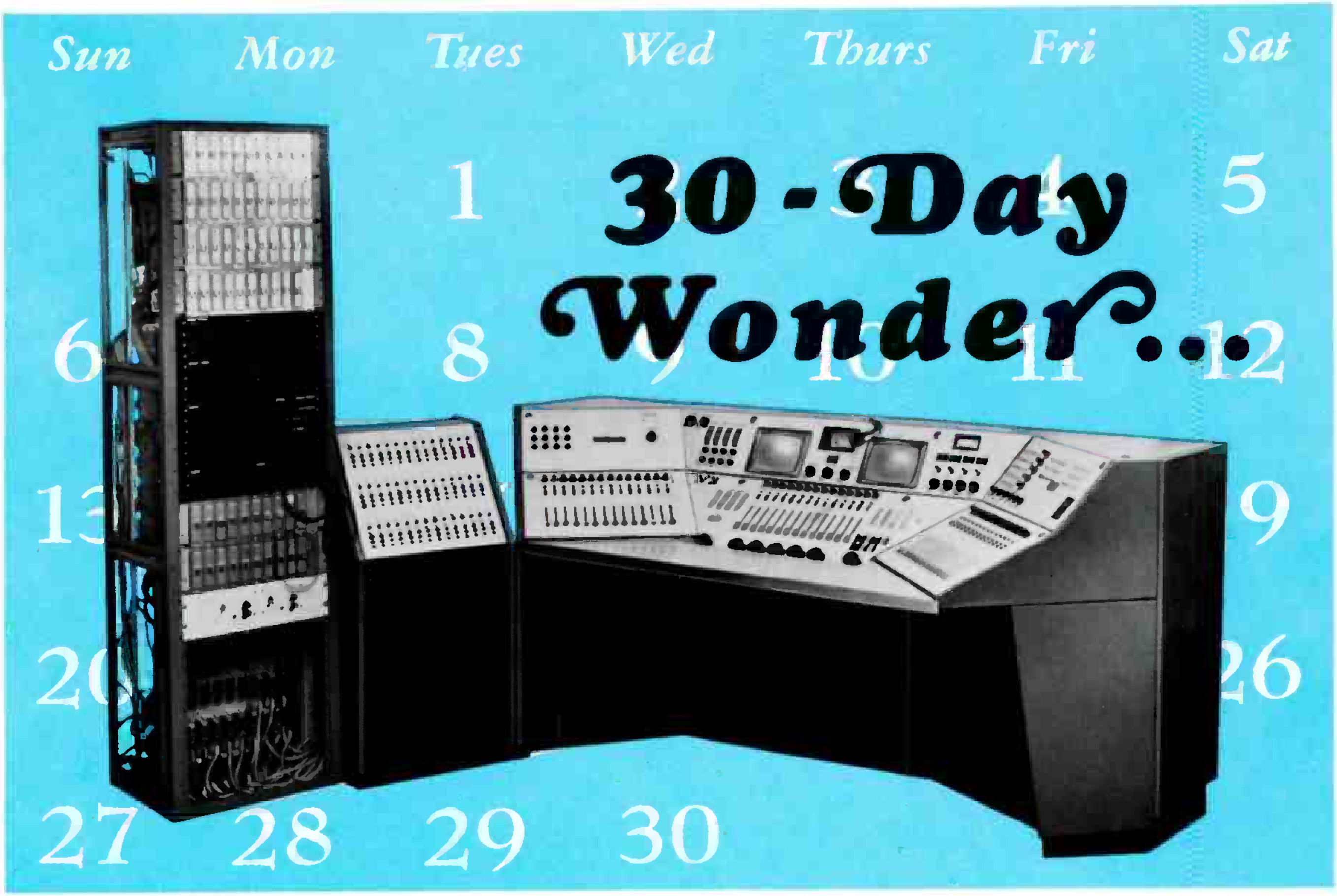

New Fairchild Integra II Remote Control Custom Audio Console!

A major TV network selected FAIRCHILD RECORDING EQUIPMENT CORPORATION, from among several of the largest broadcast equipment manufacturers in this country, to design and construct a 42-input audio mixer console. Not only did FAIRCHILD deliver a remote control mixing console in substantially less than the required time, but the network's audio engineers were so deeply impressed with the INTEGRA II console's performance and compactness that additional consoles were ordered . . . the next INTEGRA II console was constructed and delivered in thirty days. These consoles were installed and in operation within a matter of days after delivery.

NO AUDIO IN THE CONSOLE

There is actually no audio in the INTEGRA II console, with the exception of the audio lines assigned to peripheral effects equipment such as effects equalizers, VU meters etc. The INTEGRA II audio system, by removing the audio from the console area, eliminates the need for audio equipment to be located adjacent to the control area, thereby providing far greater design latitude.

THE SECRET OF FAIRCHILD's INTEGRA II CONCEPT

The secret of the speed in which FAIRCHILD INTEGRA II consoles are constructed is inherent in the advanced modularized solid state design. By combining several audio functions – amplification, attenuation and switching – in a complete system of plug-in cards, the construction of simple or complex consoles is implemented in short periods of time. In addition, a considerable savings in space and cost is also achieved.

The FAIRCHILD INTEGRA II remote control audio consoles and components are more compact, easier to install and maintain; are far superior in performance and reliability.

IF YOU ARE A PROGRESSIVE BROADCAST OR RECORDING STUDIO, with an eye to the future, look to FAIRCHILD for INTEGRA II consoles or components today. Write for complete details and brochure.

FAIRCHILD RECORDING EQUIPMENT CORP.
10-40 45TH AVENUE • LONG ISLAND CITY, NEW YORK 11101

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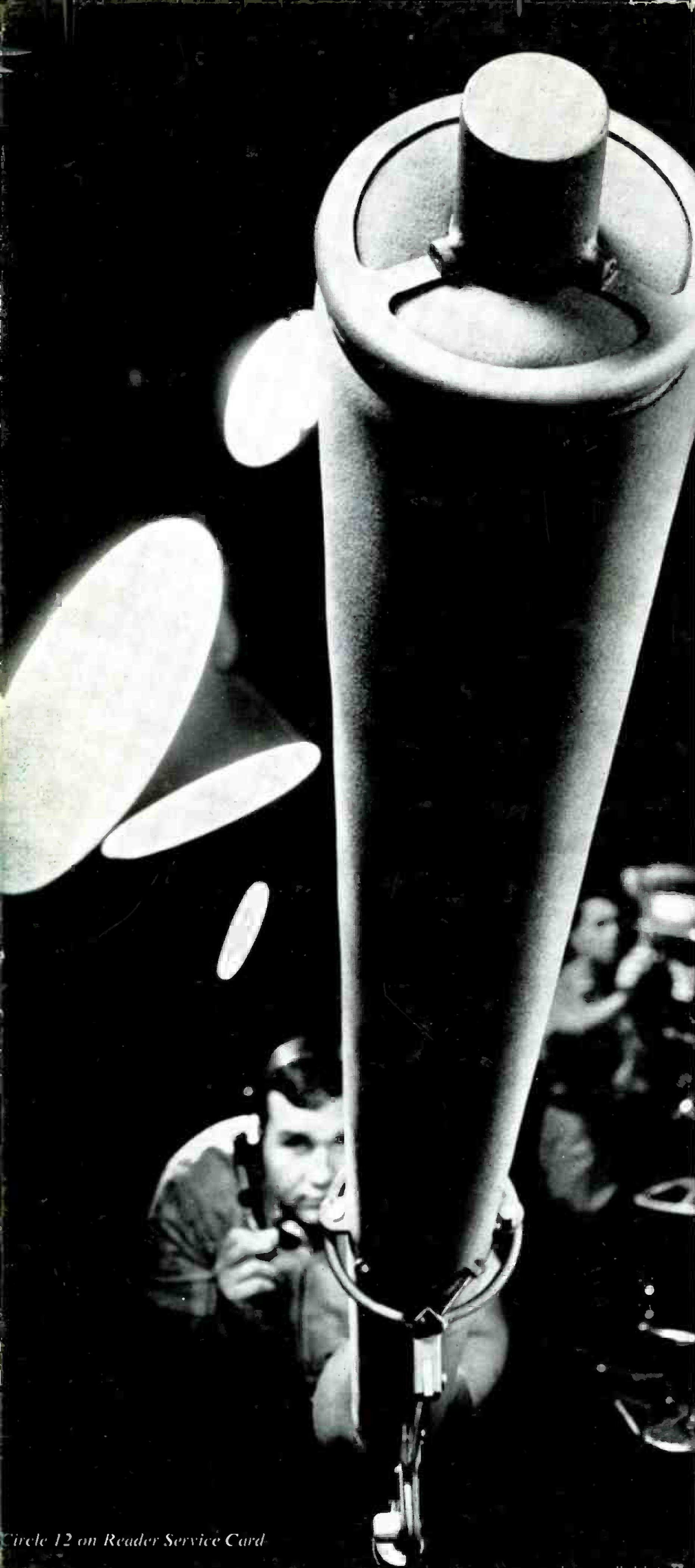


Circle 12 on Reader Service Card

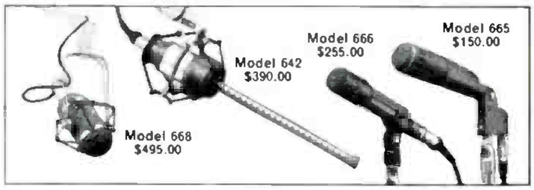
How does this 7 FOOT MONSTER help solve your sound problems?

The giant microphone shown here is the biggest microphone in captivity! The Model 643 is also the most directional microphone sold today. It helped E-V win the first Academy Award for microphone design in 22 years.

But beyond this, the 643 has been one of our most effective field research tools, offering a far-reaching insight into the nature of directional microphones, and their applications.

An obvious result of 643 research is our unique Model 642. Same E-V Cardiline™ principle*, but only 18 inches long. It reaches up to twice as far as any other broadcast unidirectional microphone to give you better long distance pickups than were dreamed possible a few years ago.

And this same basic research stimulated the development of our new Model 668 cardioid microphone. It uses the Continuously Variable-D® cardioid principle (a creative development from our exclusive Variable-D patent*) to provide smoother cardioid action—plus more versatility—than any other boom microphone you can use.



But let's not ignore the most popular professional cardioid microphone of all, the Model 666. Here's where the Variable-D principle got its start. And since the introduction of our seven foot laboratory, the 666—and its companion, the 665—has been further refined to offer better performance and value than ever before.

From such startling microphones as the 643, come continuing basic improvements—and the tools you need to solve your most difficult sound problems. Only E-V provides this kind of design leadership. E-V microphones in your studio will give you a big head start toward better sound. After all, we're at least seven feet ahead of everybody!

Model 643, $1,560.00. Normal trade discounts apply on list prices shown.

*Cardiline Patent No. 3095084. Variable-D® Patent No. 3115207

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