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THE SOUND ENGINEERING MAGAZINE
September 1970 75c

IC's — The Coming Revolution
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How good is the new Electro-Voice RE20 studio dynamic microphone?

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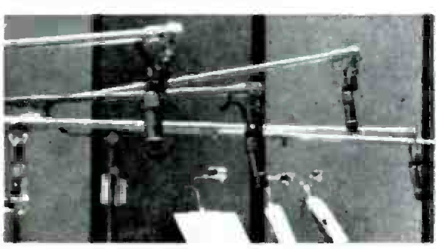

Glen Glenn put the RE20 to the test. Including days of studio experiments and actual sessions that pitted the RE20 against every type of musical instrument. Plus a searching critique by the musicians themselves. The RE20 passed every test with flying colors.

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Coming Next Month

● A special issue devoted to the exploration of compresors and limiters. We will have a round-table discussion among prominent manufacturers that should answer many of the questions you may have about compressor/limiter capabilities and performance. In addition, there will be a directory of currently-available products.

Richard Rogers has prepared an article that details the first three-way quad-stereo/regular stereo/video broadcast. It took place recently in California among two FM stations and a TV station (supplying the picture and a mono mix).

John Borwick has another European report, this time on the London Professional Audio Exhibition that was held this summer.

And there will be our regular monthly columnists, George Alexandrovich, Norman H. Crowhurst, Martin Dickstein, and John Woram. FEEDBACK LOOP columnist Arnold Schwartz will miss next month but be back after that. Coming in **db**, The Sound Engineering Magazine.

About the Cover

● The grotesque beast is in reality a tiny integrated circuit (a GE 2-watt power amplifier) that has been magnified by our camera to illustrate two stories in this issue.

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THE SOUND ENGINEERING MAGAZINE

SEPTEMBER 1970 • Volume 4, Number 9

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Letters

The Editor:

This is in reference to my article Low-Frequency Sound Absorbers, which appeared in the April 1970 issue. Two typographical errors occured in the mathematics of the paper on the bottom of page 46. The first appears at the third line from the end: instead "=w3/4 2r", the line should read "=w + 2r." The second error, which may have occured in the manuscript, pertains to the equation for V (7th line from end): instead of "f_o" (which appears depressed) in the outside factor of the denominator on the right-hand side of the equation, the term should have been "f_o^2". The curves are correct.

Michael Rettinger
Encino, California

The Editor:

Those comments in the July issue by George Alexandrovich in Audio Engineers Notebook were most pithy and realistic. He has belled the cat of skimpy audio in TV (and radio) exactly right, according to my viewpoint and experience which began in the carbon mike days of the late 20's.

Yes, radio and TV have long neglected sound, and it apparently is not improving even on FM today with the slipshod methods of so many stations. I know of one big clear channel with an excellent engineering staff which boasted of spending $20,000 on cleaning up its audio to be flat from 20 to 20,000. However, the main output thereafter was 45 rpm discs of some of the contemptible discords that are labeled "modern music." The improvement in transmission was hardly apparent with the low caliber of material.

The same paradox appears in recorded music today. We are dazzled by 8 and 16 and 24 and new 48 tracks! We give a mic to every tensil or reed. We mix down to 2 track or mono. Press on a 45 disc made of mud and manure. It is played on a $14.95 discount store phonograph by a teenager with a tin ear at maximum distortion. And the material now represents a new low point in musical creativity insofar as merit or melodic worth is concerned. Advances in sound do show up in the classical and standard output, but the vast bulk of more than 10,000 singles in a year are sheer trash. Look at the album covers today. They all look alike. The songs sound alike. They have so little to say that all the album carries are titles and timing. What a waste of sound!

There is no doubt that audio techniques are way ahead of musical content these days. Wouldn't it be nice to hear those nostalgic tunes of Wayne King, Bing Crosby, Jessica Dragonette, and Toscanini, et. al, pour out the new home systems via the improved recording skills existent today? As the present crop of regressive psychiatric caterwauling expires from sheer boredom of buyers ,perhaps the wave of the future will be music to remember. . .to whistle. . .to hum. . .to enjoy. So, George, perhaps radio and TV are smart not to worry over today's music on these 4-inch speakers.

Stephen A. Cisler
Louisville, Kentucky

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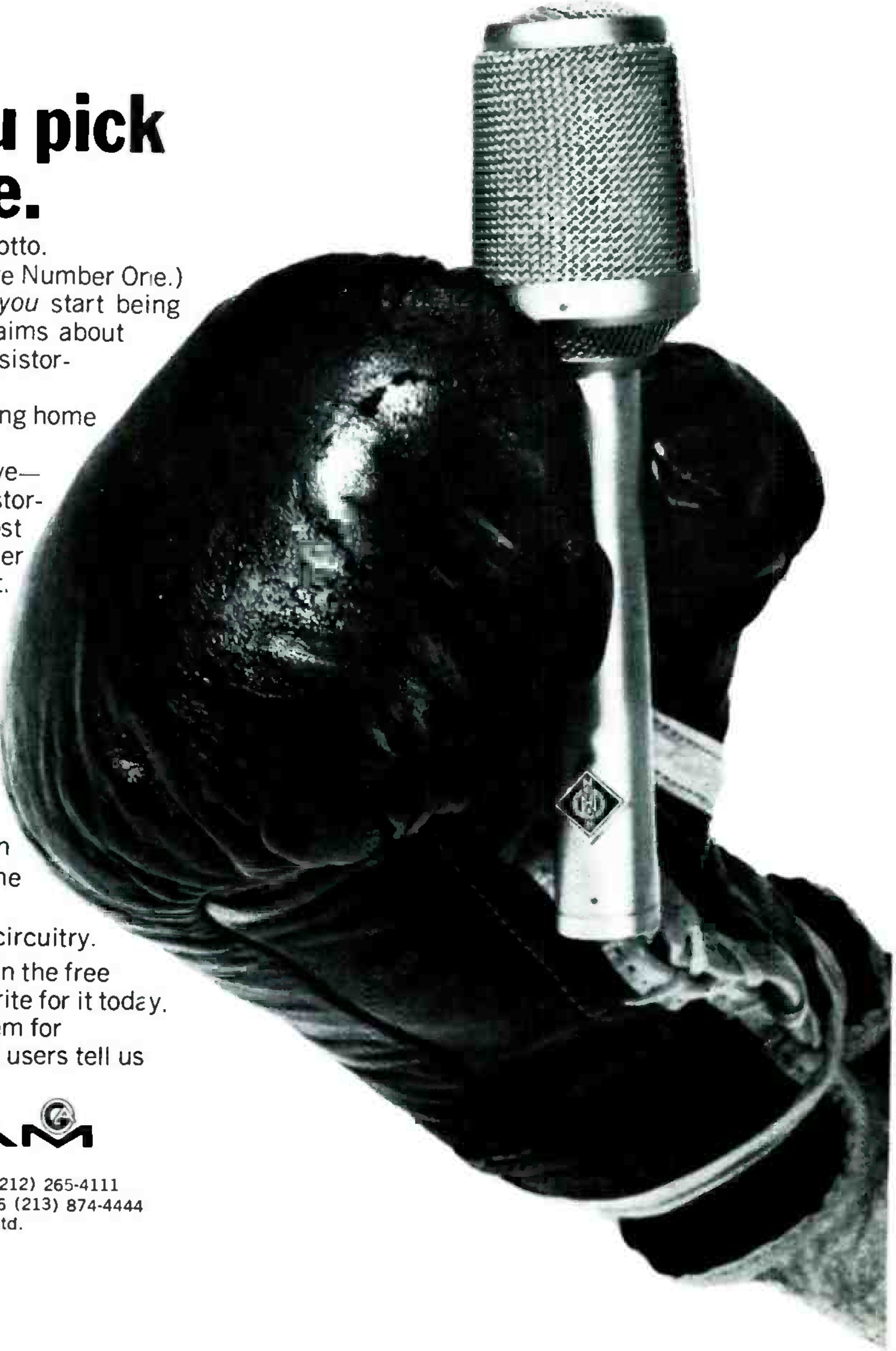

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The Audio Engineer's Handbook

GEORGE ALEXANDROVICH

S/N IN LOW LEVEL INPUTS

●Last month's subject covered measurements of s/n in the high level mixer. To complete the picture we must talk about the low level inputs and what to expect from them.

In our last month's example we obtained a s/n of 87 dB. What we did in order to obtain such a high figure was that we never allowed the signal to drop below the −40 dB level. Therefore, the difference between the signal level and the inherent noise of the amplifier was never less than 87 dB.

What do we do in the case of microphone inputs? As you know, most mics produce levels as low as −60 dBm using the standard sound pressure test. Condenser mics produce levels of −40 dBm or higher. By now we know that s/n of the system will depend on how low the signal will be allowed to drop to the noise level of the amplifier input. If the mic preamp input stage has a noise of −120 dBm and we feed a mic signal of −60 dBm, our best s/n will be 60 dB.

Sometimes this is not enough and we want a better s/n figure. There are several ways we can do this:

1. Placing the microphones closer to the source of sound.
2. Using microphones with higher outputs (condenser mics into mic preamps with lower gain).
3. Having a bridging balanced input in mic preamplifier, gaining up to 6 dB in mic level.
4. Using an expander in a mic circuit.
5. Substituting a cardioid mic for omnidirectional.
6. Improving the mic preamp input stage.
7. Improving the acoustical properties of a studio.

The experienced audio engineer will try to use every one of the above mentioned techniques in order to achieve a lower noise and cleaner sound. Neglecting any of the measures may make the difference between good and bad recording. Let us review, point by point, all of the enumerated measures.

1. Deriving a stronger mic signal by using close positioning is probably the most effective and the simplest way to improve the s/n. It stands to reason that the further the sound source is from the microphone, no matter how sensitive the mic is, the weaker are the sound waves reaching the mic diaphragm, and ambient noise pickup is more pronounced. Inverse square law states that sound intensity is proportional to the square of the distance between the mic and the sound source. It means that if we half the distance, output from our mic will increase 12 dB. Consequently, the s/n will improve by the same amount.

2. The use of a condenser or higher output mics is extremely beneficial. However, most of the time the background noise and not the amplifier noise


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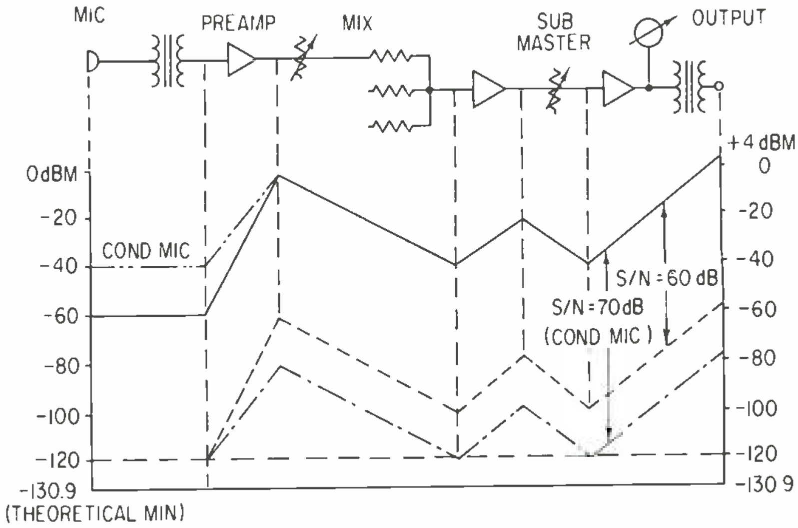

Figure 1. A diagram of level distribution.

is the domineering factor in determining s/n. Now, here is one important point. Lots of times condenser mics are padded down to prevent amplifier overload. What this accomplishes is the *lowering* of the s/n. The proper thing to do is to reduce the gain of the amplifier by varying the amount of negative feedback or adjusting the gain control which should be in the circuit after the first stage and before the output of the preamp.

3. Bridging balanced input of the mic preamp is a comparatively new approach. It offers operation with any low - impedance microphone. Microphone output is higher because it is not loaded. Some dynamic microphones change their frequency response with different loads. This is because of electrical damping the load presents to the mic. Usually, a bridging load might result in a brighter top end. If input impedance is high enough, it is possible to register peaks produced by some dynamic mics as high as 0 level (on close miking).

4. An expander or soft switch cutting the channel gain down when no signal is present, for all practical purposes may improve s/n by an order of 6-10 dB. Larger amounts of gain switching become noticeable and program material doesn't mask noise enough, contrasting it with an almost complete silence during quiet passages. This technique actually doesn't change the noise when signal is present but reduces only when it can be heard, creating an acoustical effect of lower noise. Using separate expanders on every mic channel makes gain shifting less obvious.

5. The use of a cardioid mic which inherently will reject, by a considerable amount, all signals originated outside its pickup pattern, will further contribute to the enhancement of s/n.

6. As we have mentioned before, the lowest noise we can hope for in the input stage would be roughly 3 dB above the theoretical minimum of −130.9 dBm. So far fet's are the lowest noise devices which can offer us noise levels of −127 dBm. However, don't forget you are trading in some advantages that transistors offer. These are: better stability, higher overload point, and reliability. It seems that the improvement one achieves by using fet's in mic preamps may not warrant the effort, especially if one considers that improvement is only on the order of 2-3 dB and the limiting factor in s/n relationship most of the time is studio noise, tape hiss, or other types of disturbances.

7. Naturally, if studio noise is one of the important factors—it should be considered first. Reducing air-conditioning noises, better isolation from the outside interferences, and damping unwanted studio resonances acoustically would help in achieving a cleaner signal.

We have just discussed the practical means of separating the wanted information (signals), from unwanted (noise). In conclusion we will look once more into the level distribution graph of the mic input stage (FIGURE 1) to visualize the effects of the use of the mentioned methods of reducing noise. It should not be forgotten that somewhere along the line, one could adjust the gains or levels within a system, thereby reducing s/n and nullifying your best efforts to achieve the lowest possible noise. The chain is only as strong as its weakest link; in this case the weakest link should always be the input stage of the mic preamp. To check this, the noise of the system should decrease as mic preamp is unplugged. Wouldn't it be ridiculous if the mic signal would be −50 dBm, with 70 dB s/n at the output of the mic preamp, while the system's s/n would be only 60 dB because the level was allowed to drop 10 dB below the mic level somewhere along the line. ■

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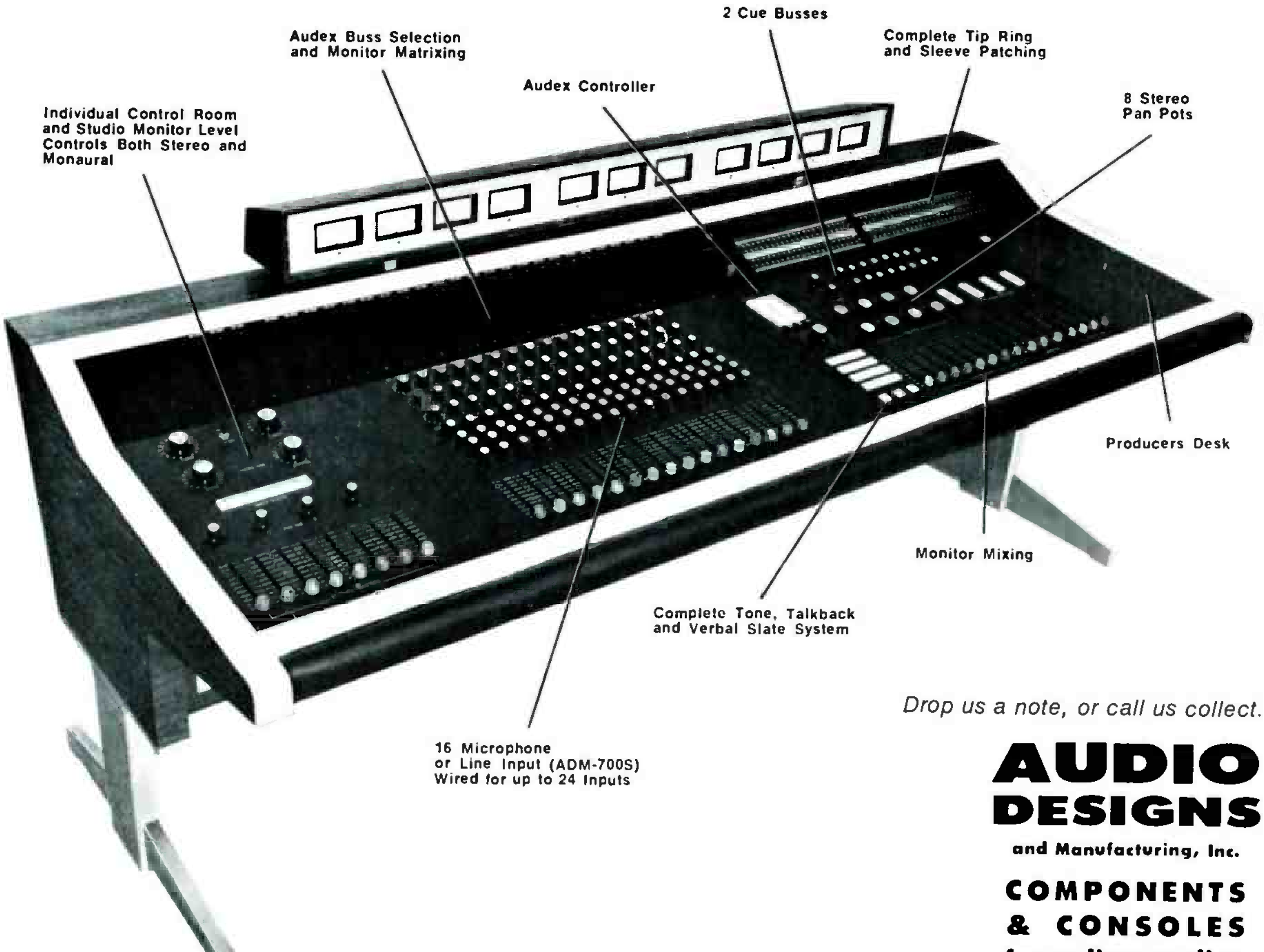

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The Sync Track

JOHN M. WORAM

ECHO - REVERBERATION SYSTEMS

● In doing a little experimenting in four-channel recording and mastering work, I've had occasion to get involved with echo and reverberation systems. By its nature, the four-channel format often requires echo/reverb facilities more complex and/or flexible than those required for regular two channel work. However, even if you're not involved in four-channel (yet), some of the techniques are suitable for two-channel work, and may represent some improvement in sound quality. Actually, there's nothing terribly new in these next few paragraphs — they're more of a rehash of techniques that may have become overlooked in the multi-track forest.

Now that I can't be blamed for talking about something you already know all about, let's lay on a few definitions,

GENERAL

Echo: A repetition of sound
Reverberation: A re-echoed sound
Delay: To postpone to a later date
Decay: Progressive decline

SPECIFIC

Echo: one (or a few at most) regularly spaced repetitions of an audio signal.
Reverberation: *many* repetitions, becoming more closely spaced (denser) with time.
Delay: the time interval between a signal and its echo(es).
Decay: the time it takes for a sound to die away.

In recording, these terms; *echo-reverb*, and *delay-decay*, are often interchanged, resulting in some confusion. At least for the purposes of this column, echo and delay will refer to the output of an auxillary tape machine or similar device; reverberation and decay refer to the output of a room or artificial device such as a steel plate, or spring.

One characteristic of a natural echo is that it does not originate at the same point as the direct sound. The term *echo* usually implies a signal displaced in time *and* space from the original sound.

In a two-channel program, there is not too much one can do about the space factor, since we are more or less limited by the distance between the two speakers. If the direct signal is located in the center, the echo might be panned left or right. However, then the total signal may sound one-sided. If the echo is also placed in the center, one loses most of the sense of space. And if the direct signal is put on say, the left, with the echo on the right, a ping-pong effect may result.

Since a reverberation device produces (or simulates) a series of echoes, it usually will sound more natural than a single echo. Unfortunately, many artificial reverb devices have only one output, thus restricting their effectiveness, since another important factor of natural echo or reverberation is its omni-, or at least multi-directional nature. If your reverb units have only one output each, it may well be worth feeding two units from one echo send line, and returning the outputs to left and right. Then, regardless of the location of the direct signal, the reverberation will sound a little more spacious. The two returns, panned left and right, are not the same as a single return in the center. A single return is just another mono point source, and although the two returns are, of course, both point sources too, they are at least somewhat different from each other, and will create more of a feeling of space.

A reverberation device does not delay the signal passing through it, so for further improvement, a tape delay system is often inserted before the reverb device. By varying the delay time, different sensations of listener-to-source distance may be approximated. Generally, the closer one is to a direct signal, the relatively further away (delayed) are the reflected signals. Thus careful adjustments of delay time may also enhance the special perspective, especially if the complete program contains more than one delay time, so that the ear references one against the other. Even the contrast of a delayed feed on some signals versus a direct feed on others may be effective.

Optimum results are probably achieved by a combination of echo and reverberation. That is, one or more delays are combined with the direct sound, and an additional delay is used to feed a reverberation chamber. Often, the delay device outputs can be fed back into its own input to produce a long series of delays that decay exponentially.

Many studios cannot afford the luxury of assigning a room as a reverberation chamber. Generally, a recording studio is in a high-rent district where

every inch has to be accounted for, and a few thousand cubic feet set aside for reverb can be an expensive proposition. However, four-channel may encourage some re-thinking on this point. A natural reverberation chamber may easily be equipped with more than one speaker and microphone. Stereo rooms often sound conspicuously better than other devices. It follows that more than two outputs might be an advantage for four-channel work. Four cardioid microphones with differing frequency responses could be fed to the four tracks, thus giving different, though related, reverb information at each speaker. Presumably, no more than two speakers within the reverberation chamber would be required.

Getting back to two-channel, a good general purpose echo/reverberation system can be assembled using two echo send lines, which I'll label left and right. A direct signal panned to the left should use the right echo send, and *vice versa*. The left echo send line feeds track 1 of an auxilliary four-track tape machine. The track 1 output is returned to the right program track, and also to the aux. track 3 input, whose output feeds one side of a stereo reverb device. In the same manner, the right echo send line is fed to tracks 2 and 4, and the other side of the reverb device. Lacking two inputs, you could combine the track 3 and 4 outputs. However, the reverb device should have two outputs, even if this means using two units. In that case, the inputs should be mixed together, perhaps in unequal proportions. The reverb device outputs are panned left and right.

This system presupposes two extra echo-send lines. If these are unavailable, you might try feeding the auxiliary tape machine directly from your left and right program buses, although this means that your entire two-track mix is sent to the echo/reverb system. Depending on the nature of the program, this may or may not be effective.

Or, if you have unused tracks (it'll never happen) on your multi-track master, you could prepare, and record on these tracks, an echo program prior to your final mixing session. If you try this, keep in mind that when you play back the complete tape, including the newly-recorded echo program, the echo program will have a built-in delay corresponding to the distance between your record and playback heads. If this delay is unsuitable, you can either play the echo program from the record head, or use the record head when preparing the echo tracks.

Then again, you could tell the producer that all the in crowd aren't using echo or reverb any more and that the new sound is dry all the way. He probably won't believe you, but after you've shot a few hours trying to get some of the above systems to work, he may be ready to surrender. ■

Theory and Practice

NORMAN H. CROWHURST

● Having covered the questions of speaker units for low frequencies, midrange and high frequencies, and arrangements for good stereo, one recurring question remains, connected with the speaker end of a system. This is what kind of crossover to use and where.

If a two-or multi-way speaker system is located at the other end of a connected cable from the amplifier that drives it, simple economics of connection suggest that the electrical crossover should be part of the loudspeaker assembly, so only 2 wires are needed from the amplifier to the speaker. But assuming this is decided, a recurrent question then concerns series versus parallel type crossover networks.

A long time ago, somebody said that the parallel variety was best, for some reason or other, which now seems to have become lost in the past. Because I have said that, some reader will probably dig up the reference(s) and let me know, in which case I will be

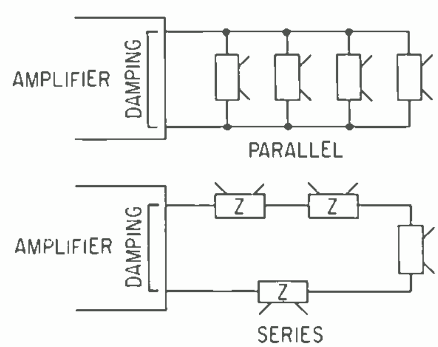

Figure 1. A comparison between parallel and series connection of speakers at an amplifier output to show the reason, based on damping, usually expressed for preferring parallel connection.

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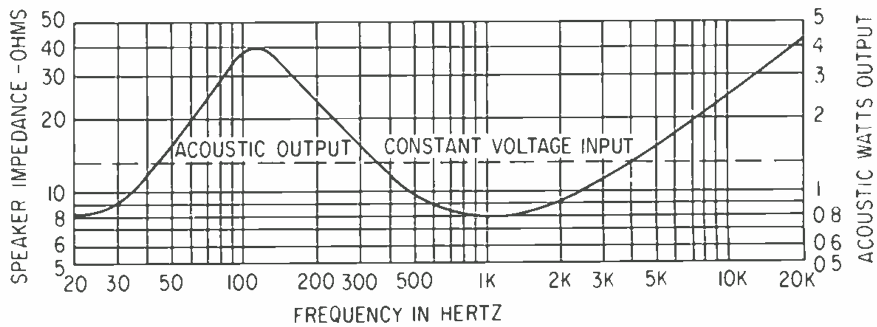

Figure 2. The effect of damping factor on frequency response. The solid curve represents a typical impedance curve. The dashed line is an ideal constant-voltage frequency response (it may not be this good). The solid curve also represents the frequency response when fed constant-current, or from an amplifier with a low damping factor.

pleased to pass on more precise information about this point.

However, when damping factor became a fashionable topic (and I believe the series *vs* parallel crossover discussion started *before* that) a more obvious reason came to the fore. Parallel connection means each unit—low and high—gets the benefit of the amplifier damping factor, without having the other unit in series with it (Figure 1).

This may have been a legitimate argument for multiple speaker units, used without benefit of crossovers. With all the speaker units active in the same frequency range, impedances reflected by any of them could affect others in the group. Thus putting speakers in series meant that each unit had others in series with it, to act as impedance between it and the low source impedance provided by the amplifier's high damping factor (assuming the amplifier *had* a high damping factor).

However, even in this relatively simple situation, the argument had some limitations. If all the units are identical, which often is the case, they also have sensibly identical performance. In addition, they often possess acoustical coupling between their outputs.

Whether or not acoustical coupling helps, a number of identical units in series will produce n times their identical impedance characteristic, while the same units in parallel will produce the same impedance characteristic divided by n.

The theory of damping factor usually set forth is that the low source resistance represented by a high damping factor allows current damping of voice-coil overshoot: it gives more precise control of voice coil movement, in accordance with the audio output voltages.

The argument for parallel connection is based on a concept of the amplifier's internal impedance acting like a short circuit strap across all the voice coils connected in parallel, which cannot act directly across each one, when they are connected in series. This argument makes the mistake of considering one unit at a time, assuming all the other units are inert impedances meanwhile.

If the units are sensibly identical, then each will want to produce sensibly the same overshoot, in response to the same initial audio input, all at the same time. Connecting them in series will cause all the motional voltages to add. So the short-circuit effect of the amplifier damping factor will cause just as much current to flow through all the voice coils in series, as would flow through each, if they were independently short-circuited directly, or in parallel.

While high damping factor undoubtedly can improve transient performance, its more obvious effect is on apparent frequency response. A speaker's frequency response is predicated on constant-voltage input to the voice coil.

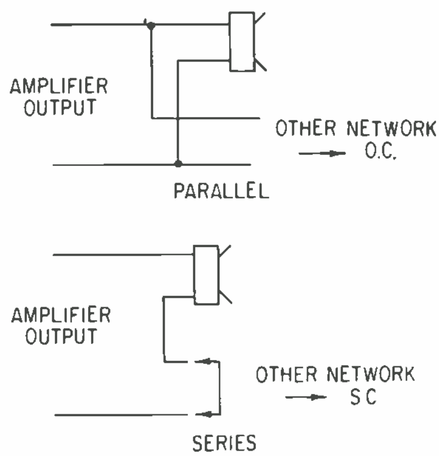

Figure 3. The actual effect of parallel and series networks does not make much difference to the coupling of damping factor to each speaker unit within its own frequency range.

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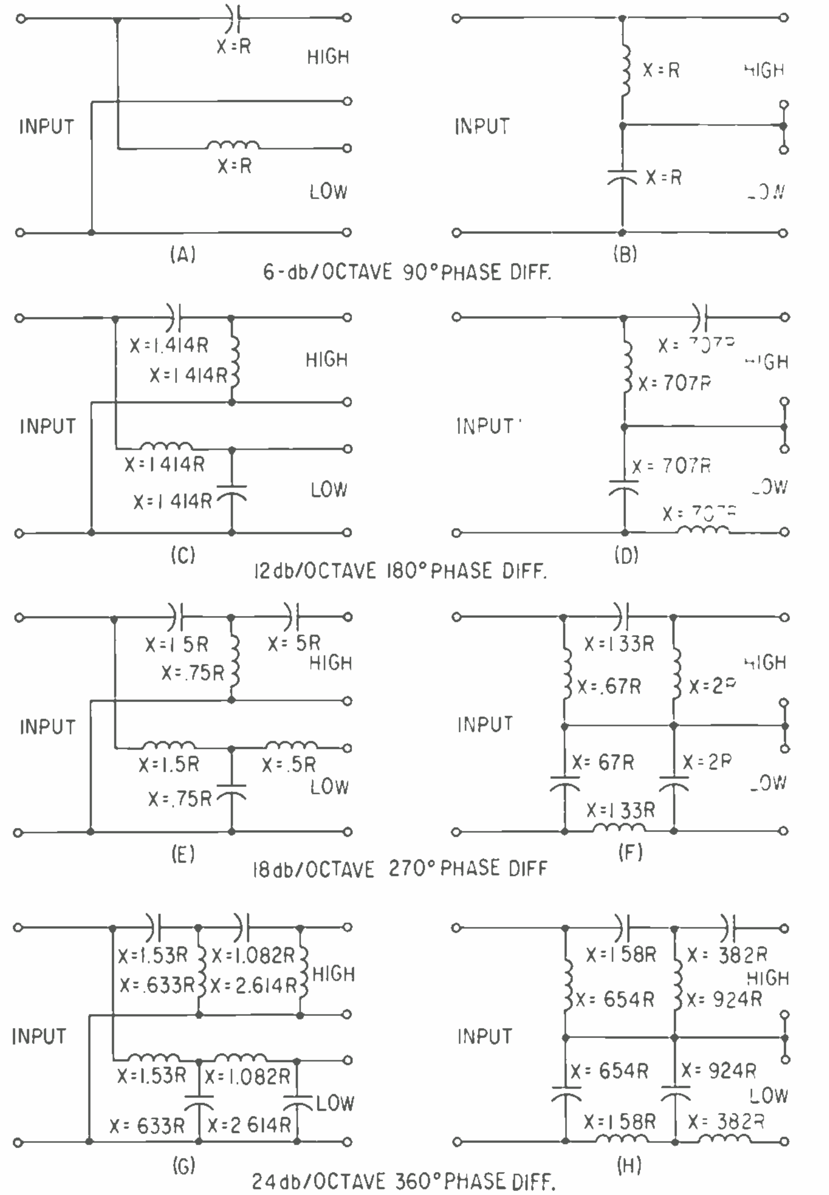

Figure 4. These are possible circuits, using from one to four reactance elements in each section to give constant resistance performance. Reactances are given in terms of terminating resistances at the crossover frequency.

A low damping factor (high source resistance) makes the amplifier output more closely approximate constant-current output, with changing frequency of constant input level.

Thus points on the speaker's impedance curve where impedance rises, notably at the resonance and the higher frequencies, will get considerably more output compared to that at other frequencies, when the speaker was designed to be sensibly flat for constant-voltage input (Figure 2).

This action will be similar, provided the amplifier is properly matched whichever way they are connected, whether the individual speaker units are connected in parallel or in series. Usually the best choice is dictated by what will most readily match the amplifier's output. Putting them in series produces a higher impedance than the individual unit impedance, in parallel lowers the resultant connected to the amplifier.

But back to the crossover question, where units are not identical, and they are not both (or all) virtually connected to the amplifier at any particular frequency: each speaker unit accepts its own frequency range, with little overlap between the units' acceptance of audio power as frequency changes.

When a parallel-connected crossover is used, the units not receiving audio power at any frequency present an impedance that runs higher than the matched impedance, and thus does not materially draw current from the amplifier's output. The amplifier's source resistance is connected almost exclusively to the speaker unit that is working at that frequency.

When a series crossover is used, the section feeding the units not receiving audio power goes to a low impedance, September, 1970

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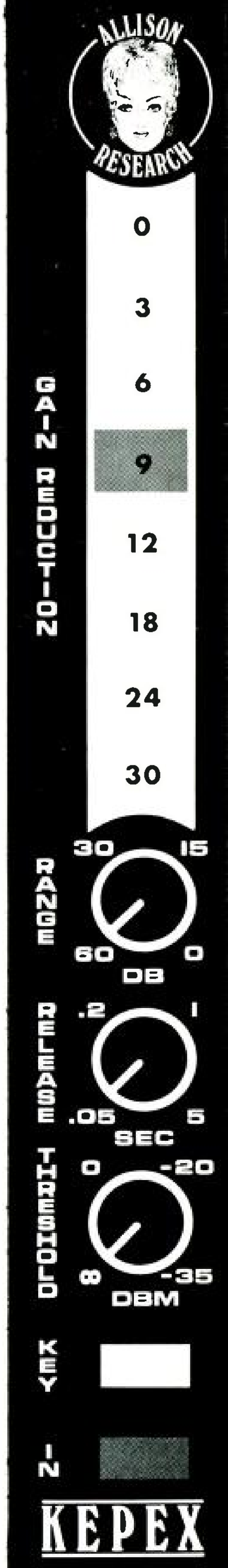

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and thus bypasses them, so the amplifier still connects virtually directly to the units receiving power at a particular frequency. So again, the difference is much less than the theoreticians who made a big case of it would suggest (FIGURE 3).

Choice between series or parallel connected filter sections may be based on a more practical factor than the theory we have just discussed. All filters are designed on the basis of feeding a resistive output impedance at each network output. Under that hypothetical condition, either type of crossover reflects a single impedance at the input, equal to the impedance connected to each output as a load.

The practical deviation from this ideal comes in the fact that loudspeakers do not ordinarily possess pure resistance impedances. Notably the dynamic type possess an inductive characteristic at the upper end of their useful response. This is taken into consideration in taking their frequency response, by feeding from a constant-voltage source.

But a correct constant-voltage source does not eliminate possible problems with the performance of the erossover, due to the fact that its networks may not be correctly terminated at their outputs. In particular, the low-frequency unit will often present an inductive load at the crossover frequency.

This can be compensated for by choosing a network that includes a series inductance element at the low-frequency output, which is true of parallel-network configurations (A) and (E) of FIGURE 4, and of series configurations (D) and (H). The effective voice-coil inductance can then be subtracted from the design value for the network used.

As these networks provide different roll-off rates, the choice of configuration may depend on the rate of roll-off desired. For 6-dB/octave and 18-dB/octave, parallel networks suit best. For 12-dB/octave and 24-dB/octave, series networks suit best.

This brings in two more practical factors: roll-off rate and phase difference. Roll-off rate is usually chosen to provide the best separation or merge (according to how you view it) between the units. On the one hand, deviations in the response of the unit going out of use should interfere as little as possible with the one in use, which calls for separation, so that each unit dominates as definitely as possible within its own frequency range.

On the other hand, where one unit takes over from the other—at crossover frequency—both need to work together, or merge, so there is not a discontinuity in the frequency response at this point. This leads into the phase problem.

Electrically the output from the crossover maintains a constant phase difference, for each type of network. For (A) and (B) of FIGURE 4 giving 6-dB/octave roll-offs, the constant phase difference is 90 degrees, or quarter of a wave. For (C) and (D), the 12-dB/octave networks, the constant phase difference is 180 degrees, of half a wavelength. For the other pairs, the differences are 270 degrees or 3/4 wave, and 360 degree or a full wave, respectively.

For outputs from the two units to merge acoustically, they need to be in phase. Most multiway units aim to be coplaner, which means their diaphragms are in the same plane for this to work correctly, both diaphrams should move together, in phase, at the crossover frequency.

The only way this can be achieved with coplanar units is to use a 12-dB/octave or a 24-dB/octave network. With the 12-dB/octave type, the units are connected antiphase, and the shift in the networks corrects it. For the 24-dB/octave type, they are connected in phase, with a whole period difference.

For each of the other types of crossover, the diaphragms should have a quarter-wave displacement at crossover frequency, so the wave from one unit has time to travel to get in phase with the wave from the other. For the 6-dB/octave crossover, the high-frequency unit should be mounted quarter of a wavelength behind the low-frequency unit. For the 18-dB/octave units, the low-frequency unit should be a quarter-wave in the rear.

Actually, the positions can be reversed from those rules, merely by reversing the connections of one voice coil. This will result in the waves generated by both units at crossover frequency emerging in phase.

That about covers the matter of using electrical crossovers with multiway speakers. We have avoided the question of how important are the phase shifts that occur between different frequency components of the program signal, or of the introduction of Doppler distortion by having the same unit handle too wide a frequency range. There are many angles to this over-all question.

This discussion was based on the assumption that we had decided to use multi-way units, for whatever reason, and then we tackled the question of which way to do it best. The one thing we have not tackled is the choice between electrical crossovers at the output, directly at the feed to the speaker units, which we have treated here in some detail, and using frequency dividers at an earlier stage, with separate power amplifiers to feed each speaker unit its frequency range separately.

So we still have more to discuss in this same general subject area. ■

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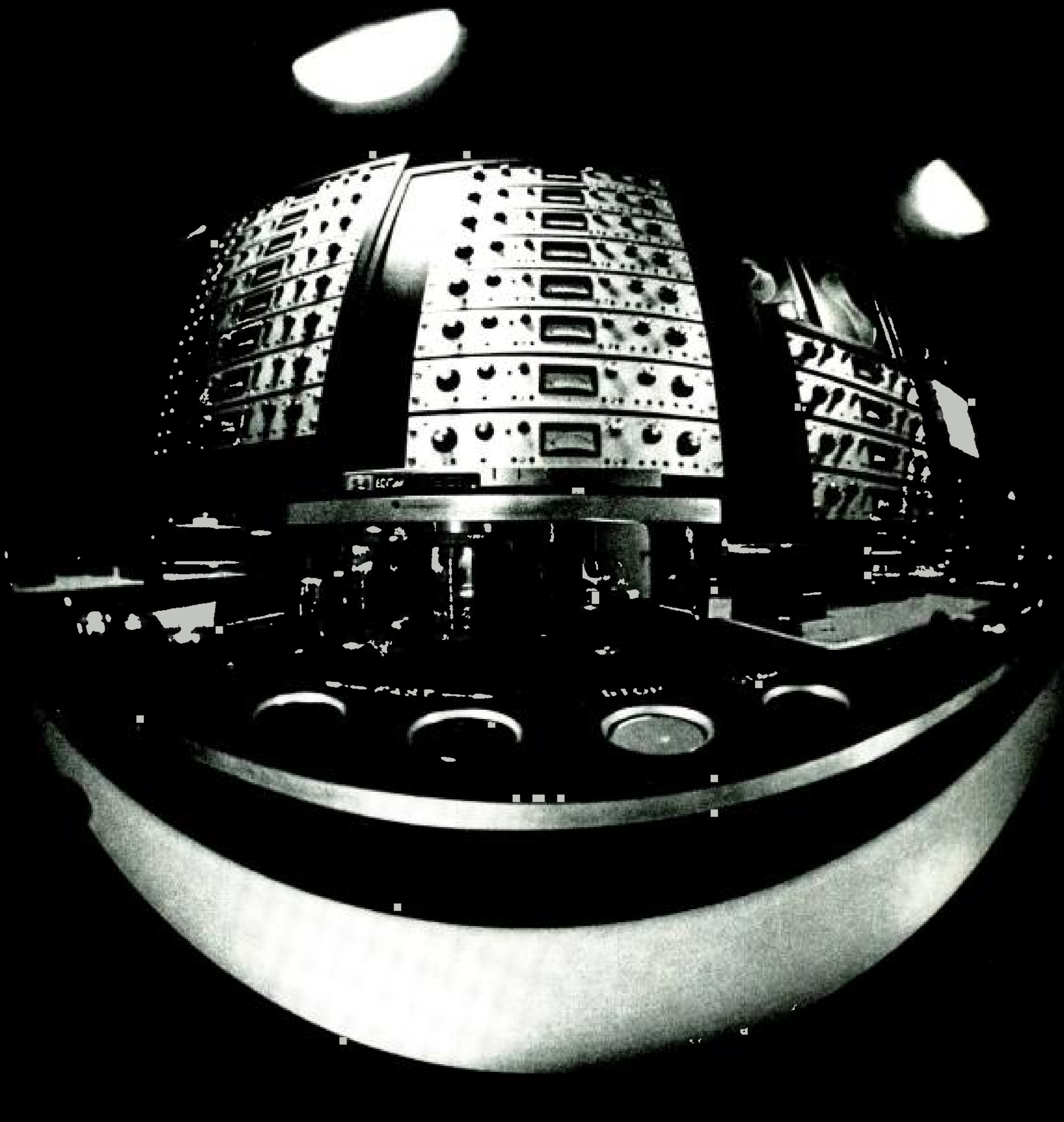

Sound with Images

MARTIN DICKSTEIN

THE NAVA CONVENTION AND EXHIBIT

●Within the past two months, audio-visual dealers, system designers, users, and anyone else interested in seeing and hearing about the latest in concepts and equipment in this field, had the opportunity to attend either of two or both) conventions and exhibits held in the East.

The first was held in Washington, D.C., from July 18th-21st by the National Audio-Visual Association (NAVA) The subject: *Media '70: Education Through Communication*. The second was the American Management Association's 6th Annual Conferences and Exposition on Education and Training, held from August 3rd-6th in New York City. The information presented herein will be on the first of the conventions. Details on the later one will follow in the next column. For those who did not attend either of the conventions, we hope to offer a few interesting highlights with a view toward having you seek further information. For those who attend either or both, perhaps we will refresh or jog a memory into requesting more details or even mention something that may have been missed.

For those not familiar with NAVA, it is the national trade association of the commercial audio-visual industry. Members include dealers and manufacturers of audio-visual equipment, producers of films and other materials used in a-v work, film rental libraries, and professionals offering services in the a-v field, as well as associate members who are interested, but not commercially involved, with the audio-visual industry. NAVA, whose office is in Fairfax, Va., annually runs a convention and trade show as well as a sales training institute. Next year, NAVA plans to run its 32nd annual convention in Cincinnati, Ohio, July 17th to 20th.

Sales meetings were, of course, held by various exhibitors at breakfasts, lunches, dinners, and cocktail parties. A full program was also provided for the ladies attending.

In the address by NAVA President, Alan P. Twyman, some excellent advice was given. Mr. Twyman said: ". . .the companies who sell price alone rather than price and services, will be left behind. Our markets demand and deserve service and consultation. . . .If you. . .fail to provide your markets with the services they need, must have, and are willing to pay for, your markets are willing to by-pass you. . .If all you

Figure 1. The Honeywell Elmo 16-S projector. It has 2000 foot capacity and still and reverse direction capability. The single button under the front of the handle sets both the automatic threading mechanism and turns on the motor.

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Distortion: less than 0.5%
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do is ship and bill, who needs you?. . . The companies who survive the '70's will be the most professional in the bunch. The professionals in our industry realize *now* that they *haven't* all the answers. But what makes them professional is the fact that they will *take the time* to find out and to do something about it."

Meetings were held by the six NAVA Councils: Audio-Visual Equipment Manufacturers' Council. A-V Systems Council. Educational Materials Producers'Council. Film Council, Industry and Business Council, and Religious Council. At the Systems Council meeting, where the purpose of the meeting was to further the knowledge of members in the fields of system design, application, installation and operation, one of the speakers, Mr. Eugene Demick, picked up the thread of Mr. Twyman's remarks in expressing the philosophy behind the success of Bergen Expo Systems and Bergen Motion Picture Service, Inc., Lodi, N.J., of which Mr. Demick is President.

Mr. Demick said: ". . .we do not sell *shop facility*. We sell *ability*! . . .the real answer to audio-visual expertise lies in ability. . .not facility. . . .To be successful in our industry demands that you gain access to the best talents available. . .Don't be afraid to work

Figure 2. The television Dolly-Lite unit can be handled by one operator. Everything folds into the unit for storage.

with other people in our industry. Don't be afraid to take another company into the deal with you. . *.even if they are a competitor*. . . .Because Bergen is not afraid to subcontract work, our facility is unlimited." Mr. Demick said that his shop, which owns no sophisticated machine tools, and has a facility of only about 2,000 square feet, was the major American supplier of audio-visual systems at Expo '67 in Montreal and is now maintaining all the a-v exhibits in the U.S. Pavilion at Expo '70 in Osaka, Japan, as well as providing equipment and services to several major installations in the New York City area.

To indicate the size of the present a-v market, Mr. Hope of Hope Reports, Rochester, N. Y., indicated at the Industry and Business Council that ". . .investment in the audio-visual communication media in 1969 was up 7 per cent over 1968 as total spending for products, services, and personnel reached the record level of $1.44 billion. . .In 1969, business and industrial buying of a-v products was greater than that of schools and colleges for the first time since 1965." Mr. Hope predicted continued growth for the a-v industry and forecast a $4 billion market by 1980.

Concurrent meetings of a-v users included sessions of the A-V Conference of Medical and Allied Sciences, The 11th Annual National A-V Education Forum, The Industrial A-V Association and the Religious A-V Conference, the last named sponsored by The Council of Churches of Greater Washington, The Jewish Community Council of Greater Washington, and The Religious Council, NAVA.

The exhibit featured displays by over 200 firms, some providing only

we call our cartridge "groovy"

you should read before you laugh!

Cartridge design is no joke with our talented, imaginative Danish designers. When they say "groovy", they're talking about the Bang & Olufsen SP-12 cartridge and its high compliance, excellent frequency response, and amazing channel separation. ☐ They call it "groovy" because an ingenious, exclusive Micro-Cross mount for the stylus allows the solid one-piece diamond to float freely in the record groove, reach the most sensitively cut undulations. The result is crystal clear reproduction of every tone hidden in a record. ☐ Your hi fi dealer knows the story of Bang & Olufsen cartridges and the Micro-Cross design that is carefully created in Denmark, presently earning rave notices from European hi fi experts, and now available in the United States for the first time. Ask him about us. Or write for details now.

SPECIFICATIONS ■ **Stylus:** Naked Diamond (5×17) μ Elliptical (LP). **Frequency response:** 15-25,000 Hz ±3 dB 50-10.000 Hz ±1½ dB. **Channel separation:** 25 dB at 1,000 Hz 20 dB at 500-10,000 Hz. **Channel difference:** 2.0 dB. **Compliance:** 25 10 ⁶ cm/dyne. **Tracking force:** 1.0-1.5 grams. **Output:** 1.0 mV/cm/sec. 5.0 mV average from music record. **Recommended load:** 47 K ohms. **Vertical tracking angle:** 15°. **Weight:** 8.5 grams. **Mounting:** ½" Standard 5 Terminal connection incl. separate ground pin. Balanced or unbalanced. **Replacement Stylus:** Original (5×17) μ Elliptical (LP), type: 5430 or 15 μ Spherical (LP), type: 5429. ■ MODEL SP-12 .... $69.95

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hardware, others supplying both hardware and software, and some furnishing auxiliary and accessory services to the a-v market. To attempt to run down the list of exhibitors or to try to describe, even briefly, many of the exhibits would be impossible. So, several interesting displays were chosen at random to provide a quick and rough idea of the range of displays offered.

In projection equipment, Honeywell Photographic Products, Littleton, Colo., showed several interesting items. One, the Honeywell Elmo 16-SS projector is a 31 pound self-threading 16-mm optical sound projector with an exclusive oil circulation lubrication system which pumps oil continuously to the vital intermittent film transport mechanism. The projector uses a 24V, 250W halogen lamp with dichroic mirror and 50-mm f/1.3 lens, and comes with a 15-watt solid-state amplifier and 5-inch speaker. All parts of the projector such as the motor, blower, shutter assembly, oil pump system, amplifier, etc. are individual modular components which permit quick servicing. The unit sells for $895.00 Other projectors in the family are the 16-S (at $1,195.00) which incorporates silent and slow motion speeds, ability to accommodate at the flick of a switch self-edited magnetically recorded films, a 25-watt amplifier; and the 16-SR (at $1,295.00) which adds the ability to record magnetic sound tracks. Other projectors by Honeywell included an auto-focus/preview slide projector and an overhead projector. (See FIGURE 1.)

From Intermedia Systems Corp., Cambridge, Mass., comes a unit labeled the R-7 which is a solid-state, variable-speed dissolve for cross fading and sequencing images from two slide projectors for special effects. The unit is controllable in one of three modes: manual advance, pre-set automatic cycle rate, or external program pulse. The front panel of the unit has controls for turning on the individual projectors, focusing and advancing or reversing either projector (or both), a cycle rate control for a range of from 1 to 20 seconds, a dissolve rate control which permits a range of fast-cut to 15-second slow dissolve, and an overlap percentage control which provides a choice of dissolve from superimposition of images to cross-fade between the images. (No price was given.)

For TV, a company called TV Dolly-

Figure 3. Resembling a small television set, the See/Hear sound slide projector is provided with a handle for easy portability.

Lite Systems, Lodi, California, showed a complete unit (FIGURE 2) on wheels which provides a mount for a TV camera and monitor, a platform for a VTR unit, and five integral quartz lamps and a microphone mount on an extension boom. The 220-pound unit can be easily moved from room to room as it will fit through any standard 32-inch doorway and has a single a.-c. cord. The unit is not supplied with a camera, VTR or microphone but will hold all cameras (except the large RCA color camera) and any VTR (except the Ampex 7800 series). The unit draws only 30 amps during operation. (No price was given.)

An item introduced at the show was the MagicVision System by See Hear, Inc., Los Angeles, Calif. The unit (FIGURE 3) is a sound/slide projection system capable of projection onto a full-size front screen as well as on its own 7 x 9 inch rear screen. The synchronized system uses a standard compact cassette and a slide tray capable of holding up to 50 35-mm slides which are loaded into the unit from the front. The unit will sell to educational institutions for $350.00.

The NAVA Exhibit was also the site of the initial showing of a device for photographers and those making their own audio-visual color prints. The O/G Chroma 810 Color System by Opto Graphics, Inc., Northbrook, Ill., offers the user the ability to create, in any space that can be darkened temporarily, permanent full-color prints up to 8 x 10 inches directly from any type color slide or transparency, without water or an internegative, in less then 15 minutes. If you missed this show or want to see the device again, it will be on display at the biennial European photographic exposition in Cologne, Germany, in October. (It was also shown at the Professional Photographers of America meeting in Chicago last month.) Suggested retail at $169.95.

By selecting these items from the tremendous number of devices and services presented during the NAVA show I do not mean to imply in any sense that we recommend or approve of the equipment in any way. These few items were only chosen as an indication of the breadth and depth of the exhibitors and their displays. I do, however, wish to impress on you the need for further and continued investigation into the new equipment becoming available almost every day, as well as the novel uses to which these audio-visual devices may be put, not to mention the continually increasing number of a-v users in all phases of industry and education. The a-v market is growing in leaps and bounds (it's already over the 1 *billion* mark). In order to get and keep and continue to get your share of the market, remember the need to furnish more than just equipment if you want satisfied customers. ■

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The Feedback Loop

ARNOLD SCHWARTZ

●WHAT is the future of the phonograph record in light of the rapid growth of the cassette? In the April FEEDBACK LOOP, as a way of comparing the performance capability of tape and disc, the relative information storage capacity of each medium was discussed. This article prompted Harry Maynard, producer and moderator of a New York radio program *Men of Hi Fi* to devote a program to the question, *Is the Record Still King*?. A transcription of that program appeared in the June issue of the magazine *FM Guide*. More recently I received a letter from Andrew G. Petite, National Sales Manager of Advent Corporation, in which Mr. Petite states that he would like to "rebut [my] argument for the disc". Last February Advent Corporation held a demonstration at which a cassette and a commercial disc version of the same musical selection (derived from the same master tape) were compared. Stanley P. Pressman, Advent Marketing v.p., stated at that demonstration that, "We think the cassette can now approach the very best records . . .and in 6 months the cassette could surpass the record".

In an effort to achieve wider acceptance, and to boost sales, cassette manufacturers may make claims that exceed realistic engineering appraisal. As I had suggested in this column last April, each new form (in this case the cassette) seems to develop its own market in the areas where it is most suitable, without necessarily obsoleting older forms (in this case the disc). New ideas and products have usually enlarged the audio field, and provided the consumer with a wider selection for his entertainment dollar. Nevertheless, there may be a feeling in some quarters that the advent of the cassette spells the doom of the disc. It is my contention that disc, having been around for ninety four years, is too readily taken for granted. The disc is the *quality*, low-cost playback medium which requires relatively simple and durable playback equipment. I predict that the entry of the cassette into the audio market will not put an end to the continued expansion of the disc.

I would like to discuss one of the arguments Mr. Petite makes in his letter because it points out the pitfalls in making comparisons of disc and tape. In the April FEEDBACK LOOP a somewhat roundabout method was used to compare the two; that is, by comparing the information storage capacity of a 33-1/3-disc with a 15 in./sec. reel-to-reel tape on an area and volume basis. We found that the disc had a greater storage capacity either way, and hence the capability of higher playback quality. In his letter, Mr. Petite compares the 1-7/8 cassette with the 33-1/3 disc. He states, "From the standpoint of information storage, the area required for one second on the casette is 1.875 in./sec. x 0.075 inch =0.141 square inch, compared to 0.077 square inch for the disc. The volume is 0.141 $inch^2$ x 0.0007 inch (the total thickness of backing and oxide coating of a C-60 cassette . . . .) =0.0001 cubic inches for one second of playing time vs. 0.0019 cubic inches for a disc, or 1/19 the volume."

Let's start at the volume end of the argument, since by Mr. Petite's calculation, and despite the reduced tape speed, the area requirement is still in favor of the disc. The typical disc is 0.050 inches thick, but the disc is a self contained playback device-that is, the disc itself can be played. A cassette, on the other hand, consists of reels and a housing, in addition to the tape itself. Actually, the disc needs little more than 0.005 inches for the information storage, the remainder of the thickness is used to make the disc rigid so that it can be used on playback turntables, and not break during handling and storage. The additional thickness of the disc corresponds to the plastic reels and housing of the cassette. We would probably find that there is enough of this plastic in the cassette to press a complete record.

A second problem with Mr. Petite's comparison is the use of compression. The cassette in question was recorded through a single-band Dolby unit, and played back through a complementary Dolby unit. The disc was an off the shelf item, and was played back through standard playback electronics. A disc encoded with the Dolby system, and then played back through a complementary expansion device would show significant improvement over a standard disc. Interestingly enough, Sanford Drelinger, quality control director of Vanguard Records, told me that Vanguard has been experimenting with this type of disc. There is little doubt that such a disc would yield a 6-10 dB improvement in signal-to-noise ratio- at the "cost" of additional circuitry in the playback system, and of having a disc that is not compatible with standard playback equipment.

A third and basic point is brought to mind by Mr. Petite's letter, although Mr. Petite did not discuss it in his letter. Do we get more information on a recording medium by reducing the speed, or by reducing the track width?

In cassettes and eight track, there are two methods of storing the music in an acceptably small package; reduction of speed, and reduction of track width. Let us see what the effect of each of these reductions has on bandwidth, signal-to-noise ratio, and maximum level. Each time we reduce the tape speed by half, the bandwidth is reduced one octave; a reduction from 15 ips to 1-7/8 ips means a loss of three octaves—assuming all other things remain the same. If 15 kHz was originally the highest frequency reproduced, then the highest frequency that can be reproduced at the reduced speed would be 3.75 kHz. Extension of the bandwidth can be made by reducing the size of the playback gap. Each time we reduce the gap by half the bandwidth is extended one octave. At the same time, the recorded signal level drops 6 dB, and the tape noise drops only 3 dB; there is a net loss in signal-to-noise ratio of 3 dB. If the track width is reduced by half, the recorded signal level drops 6 dB, while the tape noise drops only 3 dB; once again there is a net loss in signal-to-noise ratio of 3 dB.

When both reduction of gap width and track width are employed, there can be a substantial drop in the signal-to-noise ratio. To counteract the increase in noise, some form of signal compression may be employed. And finally, to overcome some of the undesirable effects of compression, it is necessary to place a complementary expansion device in the playback electronics.

All this is to say that the information storage limitations of tape (and disc) cannot easily be circumvented by using slower speeds, and if we try to squeeze more information onto a given segment of tape, we have to pay the price somewhere along the line. ■

New Products and Services

MONITOR SPEAKER

●Model 4311 is designed as a control room monitor with on-axis response held within ±5 dB from 30-15 kHz. Wide dispersion provides less than a 6 dB deviation up to 45 degrees off axis at 2 kHz nor more than 10 dB at 8 kHz. Internal components include a 12-in. woofer, and high-frequency transducer with a crossover at 1500 Hz. Woofer free air resonance is 27 Hz. Above 3 kHz, a third transducer is gradually coming into effect reaching full output at 7 kHz. Up to 50 watts program material is handled at a nominal impedance of 8 ohms. The enclosure is solidly constructed of ¾-inch stock with all joints tightly fitted and glued. Over-all dimensions are 24 x 15 x 12 inches and weight is 51 lbs. shipping.
Mfr: J. B. Lansing Sound, Inc.
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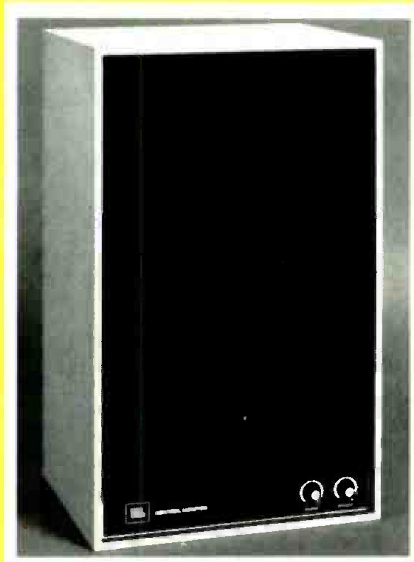

MATCHING TRANSFORMER

●The Peerless Model 15370 features hum-bucking construction, and tight coupling which results in low leakage reactance and good high-frequency response. The transformer is for use as impedance matching or bridging use in low-level 600-Ω circuits. It is hermetically sealed in a metal case.
Mfr: Altec-Lansing
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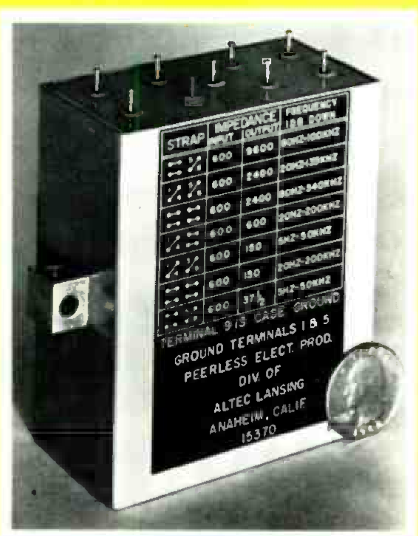

AUTOMATIC FADER

●Smooth, stepless control of any number of channels simultaneously is achieved with the press of a button on this automatic, electronic attenuator. The speed of the fade is adjustable from 0 to 30 seconds and noiseless operation is assured by use of solid-state circuitry.
Mfr: Moser Development Co.
Circle 74 on Reader Service Card

MIXING DESKS

●The MD-16 series is the latest unveiled by this company. It has as outstanding features the use of channel blocks that afford good flexibility, increased assembly and switching possibilities, as well as easy service and maintenance. The illustrated console is sixteen channel, but a variety of configurations both in and out are available.
Mfr: Philips Broadcast Equip. Corp.
Circle 66 on Reader Service Card

PORTABLE MIC MIXER

●The GCA model mixer/amplifier is designed for newsreel recording and uses transistorized circuitry to achieve its specifications. These include an equalized frequency response of 120 Hz to 10 kHz ±3 dB; unbalanced 50-200Ω mic input impedance; high level input; defeatable automatic volume control; earphone jack at 12 dBm; and a rechargeable (built-in) 12 V battery that provides a minimum of 8-10 hours of continuous service. Weight is 28 ounces with the battery and the unit's temperature operating range is −4° to 131°F. The unit has been designed to be compatible with Auricon MA-11 heads.
Mfr: General Camera Corp.
Circle 68 on Reader Service Card

SPECIAL CONVENTION SESSION: MODERN RECORDING STUDIO TECHNIQUES

At 7:30 pm on Tuesday, October 13, during the AES Fall Convention, the New York Section will present an applications session to show how a modern recording is made. A multitrack tape recorder and a multitrack console will be used in the live demonstration. Producers Max Wilcox (classical) and Steve Schwartz (rock) will explain their roles and John Woram, RCA Records, New York will explain mixing and miking with the use of visual slides and closed circuit TV, so the audience may observe the mixer at the console.

•

During the playback of a multitrack tape recorded by a 17-piece orchestra, a live guitarist will lay down his track. Following the demonstration, other professional mixers will join in a discussion and the audience will be invited to ask questions and make comments. Chairman of the session will be William E. Windsor, D.B. Audio Corp, New York.

TAPE CARTRIDGE SYSTEM

• A recent announcement tells of the introduction of this multi-deck tape cartridge machine. The unit offers economics in both size and price with its compact construction and common capstan drive. An available option is a recorder module that converts the bottom deck to record/reproduce.
Mfr: International Tapetronics Corp.
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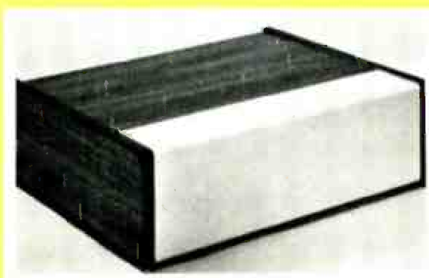
INSTRUMENT ENCLOSURES



• This new line of enclosures is highly styled and constructed of aluminum wrap-around and molded ABS end panels to provide rigidity, accessibility, and light weight. Two finishes, grey with black or white with wood grain, are available. The two basic sizes available from stock are $2\frac{1}{2}$-in. high by $5\frac{5}{8}$-in. deep and can be had 4 to 10-in. wide — or $4\frac{1}{4}$ by $6\frac{1}{4}$ and 5 to 12-in. wide. Special finishes and punched holes can be provided.
Mfr: Ten-Tec, Inc.
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INDUSTRIAL SOUND AMPLIFIER

• The Pyramid Series 7000 solid-state amplifier uses two input channels to provide separate bass and treble tone adjustments for voice paging and music. As many a ten input modules can be plugged in to provide a variety of functions ranging from mic inputs to siren alarm systems. Power modules plug in also and are interchangeable in units of 20, 40, 80, and 150 watts. A limited access door is provided. Power circuits are protected by automatic devices and a visual overload alarm.
Mfr: Private Tele-Communications
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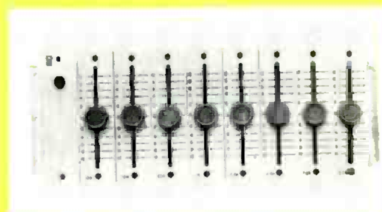
GRAPHIC EQUALIZER



• Model EQ-812 is an eight-frequency equalizer with accurately reciprocal curves and with flat frequency response at the detented midpoint position. It is an active unit with unity gain, can be switched in or out, is 600Ω balanced in and out, 12 dB boost or cut is available at each frequency. They are: 63 Hz, 160 Hz, 400 Hz, 1 kHz, 2.8 kHz, 4.5 kHz, 7.5 kHz, and 12.5 kHz. System noise is better than −80 dBm and output levels up to +24 dBm terminated can be had. In and out terminals are on a C Jones strip, the power required is bi-polar 28 V d.c. at 115 mA.
Mfr: Quad-Eight Electronics
Price: $825.00
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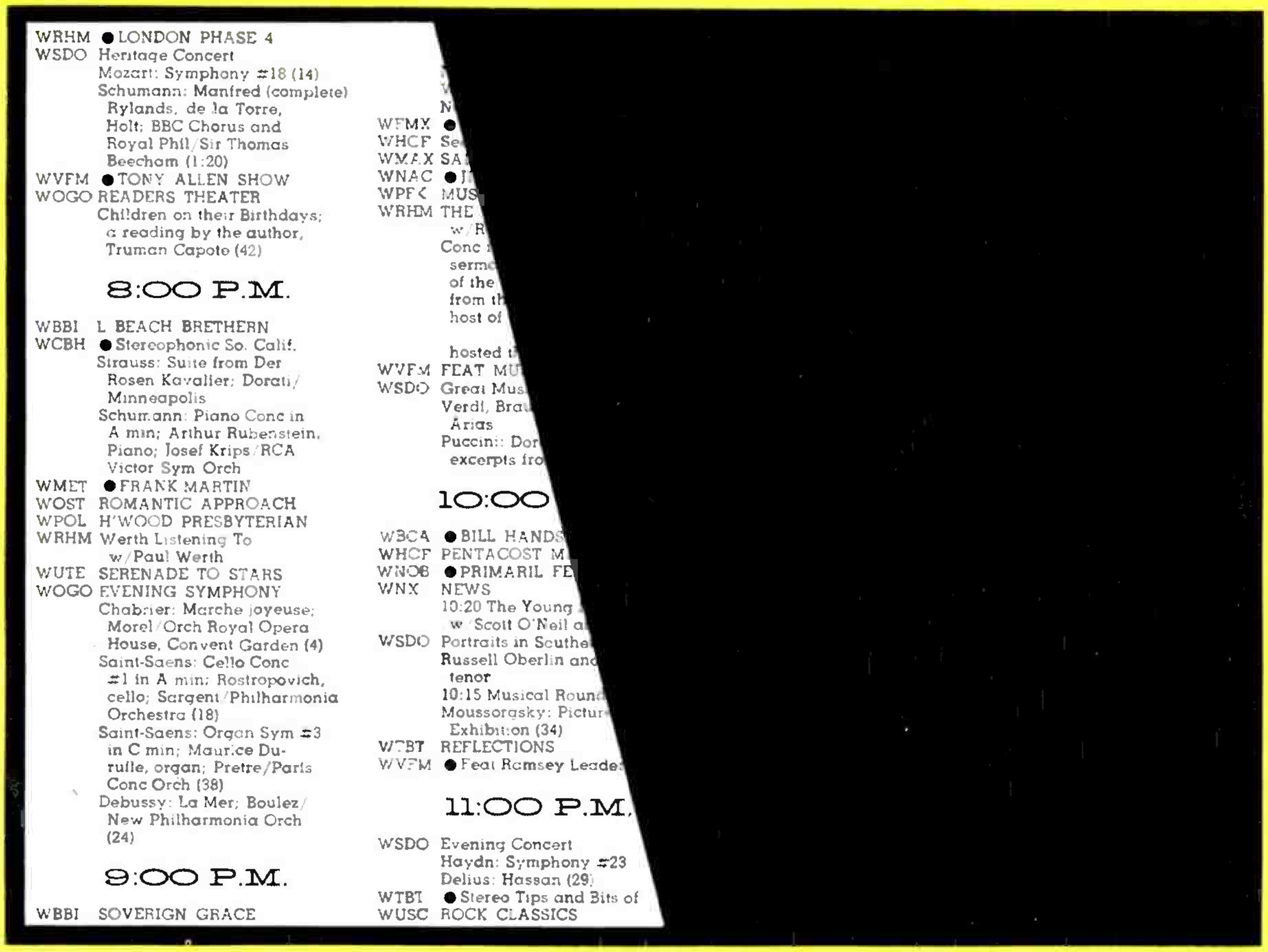
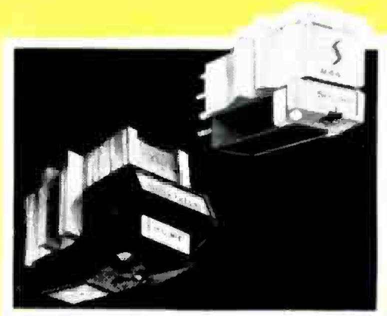
MIC TRANSFORMERS

• Miniature transformers half the size of a thimble are being made by MB in West Germany. A variety of voltage ratios and with or without mu-metal shielding is available. Model BV-16, as an example has mu-metal shielding, is 200/50k, and has vinyl encapsulation. Eight inches of cable at each end are provided for line connection.
Mfr: Stanford International (MB)
Circle 67 on Reader Service Card



Just for the record...



More FM and AM radio stations have standardized on *Shure M44 series cartridges* than any other, because of their unique combination of good sound, ruggedness, and inexpensive price. Stylus change takes only seconds. They are well suited to back-cueing. Wide choice of tracking weights from ¾ gram to 5 grams. For "high fidelity" FM stations, the *Shure V-15 Type II Improved* cartridge delivers superlative reproduction — *regardless of the level at which the record is cut.* Send for the Shure "Guide To The Conversion Of Monophonic Broadcast Facilities For Stereo Records." Shure Brothers Inc., 222 Hartrey Avenue, Evanston, Ill. 60204.

SHURE

Circle 40 on Reader Service Card

NEW LITERATURE

●A comprehensive catalog of public-address equipment is available from University Sound. It lists speakers, speaker products, and electronics. *Circle 51 on Reader Service Card.*

●The Turner Company has a 24-page catalog that describes its complete line of microphones and accessory equipment. *Circle 52 on Reader Service Card.*

●The Tape-Athon Model 1000 Recorder/Reproducer is detailed in a 4-page brochure. Rack mounted, portable, and studio console versions are described. *Circle 53 on Reader Service Card.*

●The GRT 260 tape duplicating system is introduced in a brochure and series of specifications. Both the components and their functioning together are given. *Circle 54 on Reader Service Card.*

●Altec's Acouta-Voicette process is described in a brochure. This tuning process, designed for small-room use, utilizes 24 filters per channel in a two-channel system. *Circle 55 on Reader Service Card.*

●The B & K Model 4712 Frequency Response Tracer is covered in a new product data bulletin. This tracer has a 14-in. screen display of response data in the range 20 Hz to 20 kHz (or narrower bands—or extended). *Circle 56 on Reader Service Card.*

●Wave Analyzer applications are illustrated in a 20-page booklet available from Hewlett-Packard.

It shows instrument connections on the page opposite an x-y graphical recording of the result. *Circle 57 on Reader Service Card.*

●The Model FR-10 film recorder/reproducer from RCA is described in a brochure that gives features, operating information, and specification detail. *Circle 58 on Reader Service Card.*

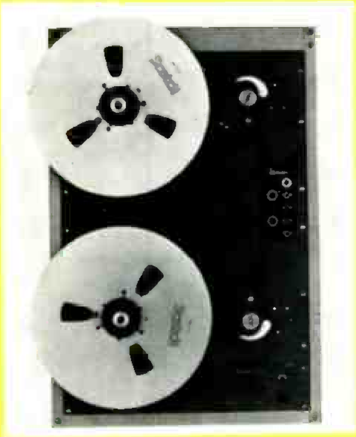
●American Electronic Laboratories offers a colorful 4-page brochure setting forth the varied capabilities of its Environmental Testing Laboratories. *Circle 59 on Reader Service Card.*

●Pre-wired power outlet boxes are described in a new Waber Electronics catalog. The 16-page booklet provides full information on more than 400 standard Waber models. *Circle 60 on Reader Service Card.*

●A new two-color brochure for the broadcast industry has been issued by the professional products department of CBS Laboratories. A complete line of audio and video products is described with specifications and prices. *Circle 61 on Reader Service Card.*

LONG PLAY TRANSPORTS

●Model TR-100 tape transport operates in a bi-directional mode by foil contacts on either end of the tape, or by manually depressing the appropriate control button. Cut-off arms adjacent to both reels prevent damage in the event of tape breakage. The unit will mount in a standard 19-in. rack and it occupies 24½-in. of vertical space. All subassemblies such as capstan motor, brake motors, control chassis, and magnetic playback heads are plug-in devices. Two standard speeds, 3¾ and 7½ in./sec., are provided; the unit uses ¼-in. tape; and it accepts reels up to 16 inches. Wow and flutter at high speed is 0.1%.
Mfr: Langevin
Circle 69 on Reader Service Card



CONSOLES

●Stock 16-track consoles are now available for immediate delivery. The 1020 series 8/16 feature 80 dB s/n, 1000% peak overload capability, and less than 0.01% distortion. Twenty mic/line inputs with equalization, eight metered program buses, sixteen outputs, four echo buses, three independent pan-pot systems, separate 16-track remix, free grouping, and patching facilities are provided.
Mfr: Spectra-Sonics
Price: $35,000.00
Circle 76 on Reader Service Card

HEAD CLEANER

●This is a new formulation designed to clean magnetic heads on both video and audio machines. It is available as both liquid and spray. Though the formulation is not revealed, it is stated to be silicone free so it can properly be used to clean guides, capstans, and pinch rollers. The spray can contains 16 oz., with a 5-in. extender tube; in liquid form both 8 oz. and 32 oz. cans are available.
Mfr: Nortronics
Price: Aerosol — $2.75; liquid — $2.25 and $7.50.
Circle 70 on Reader Service Card

●Koss offers a 16-page four-color brochure designed to explain stereophones and stereophone listening. It also functions as a catalog of the company's dynamic and electrostatic phones. *Circle 62 on Reader Service Card.*

●Elpa Marketing has a 6-pager that describes the complete line of Thorens transcription turntables and arms, as well as wooden bases. *Circle 63 on Reader Service Card.*

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An IC Line Amp-- Or is it?

WALTER G. JUNG

Useful audio products often may be found masquerading under other colors. The author describes a line amplifier of high quality that may be built with an integrated circuit not intended by its manufacturer for this purpose.

ELECTRONICS is an odd field sometimes. Quite often things turn up which seem so odd as to be unbelieveable at first glance. But with further examination and understanding comes a feeling of a "Maybe that's a good way after all" and then "Why didn't I think of that?". To illustrate just what is meant by this, let's look at a circuit which is developed with unseeming parts, but performs remarkably.

The circuit to be discussed is an audio line amplifier—a common application, nothing exotic. Before we introduce the *how*, a few numbers describing this hypotehtical amplifier might be appropriate. We want the amplifier to handle levels normally associated with a console line output; that is average levels of +0dBm and peak capability of +17dBm. This means its output impedance should be 600 ohms to match standard lines. And it would also be nice to have 2-line output capability with more than 60 dB of isolation between the two lines. A few dB of gain might be convenient, say 10 dB. The input impedance should match the preceding stage so it should also be 600 ohms. As far as distortion and frequency response go, 0.1 per cent and 20 Hz−20 kHz ±0.25 dB should be adequate. A few other niceties might want to be

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Walter G. Jung has been a frequent contributor to **db**.

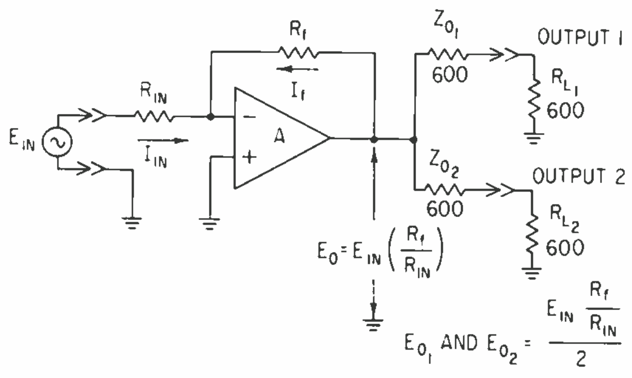

Figure 1. An opamp model of a line amplifier.

thrown in, such as single unregulated supply operation, modular IC construction, and simplicity combined with (hopefully) economy. Sound like a tall order? Before we decide, lets look at a few circuit configurations to see how such goals might be met.

Since practically everyone is talking op-amps these days we'll approach the problem from this angle. This not only makes us "in" but makes for a good amplifier too. In FIGURE 1 we see the familiar magic triangle and its associated hookup for an inverting amplifier. We won't attempt to cover all the theory associated with this handy device, we'll just be content to abide by the laws as generally set down, and use them to suit our purposes.

In an inverting amplifier such as this, the input signal is reproduced at the amplifier output 180 degrees out of phase and scaled by the ratio of R_f to R_{in}. Integral to this process is the fact that the two currents associated with the respective signals E_{in} and E_{out} sum to zero at their junction or the input node of the amplifier. And so the term arises, *virtual ground*. If this terminal is in truth an equivalent ground, then the effective input impedance will be the value of the input resistor, R_{in}.

So by now we have established two factors about our amplifier. We can set the input impedance by the value of R_{in}, and the gain by the ratio of R_f to R_{in}. But this is not all we have established. In so doing, we have generated a low output impedance voltage amplifier with its characteristic wide frequency response and low distortion, two of our prerequisites.

There remains the question of how to make the thing believe it is a 600-ohm source, or actually two 600-ohm sources, with a lot of isolation between one other. This is not so hard. Since our voltage amplifier is a zero-impedance source, a series resistor of 600 ohms will make an excellent 600-ohm impedance. The penalty here is a 6 dB voltage gain drop and a power waste, but with gain available so cheaply by merely adjusting R_f/R_{in} this is really no big thing. A purist might balk at throwing away power so extravagantly, but there is an important advantage to this technique. The isolation between the two output terminals is directly proportional to the ratio of the 600-ohm resistor to the output impedance of the amplifier. The lower the amplifier Z_o, the better the isolation. And the 600-ohm resistors buy inherent short-circuit protection against shorts on one line affecting the other.

Now we really have quite a few things known about this amplifier-Z_{in}, Z_{out}, gain, and some generalities about distortion and frequency response. To come closer to our actual circuit, we'll have to begin talking about some hardware.

We'll start by looking at the power requirements. A +17

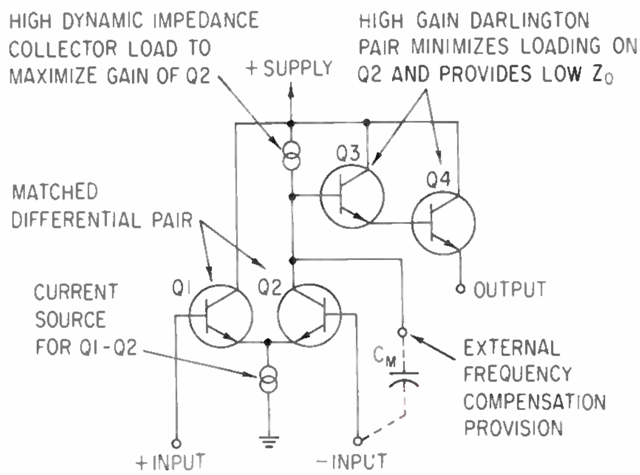

Figure 2. The circuit configuration for an IC audio amplifier.

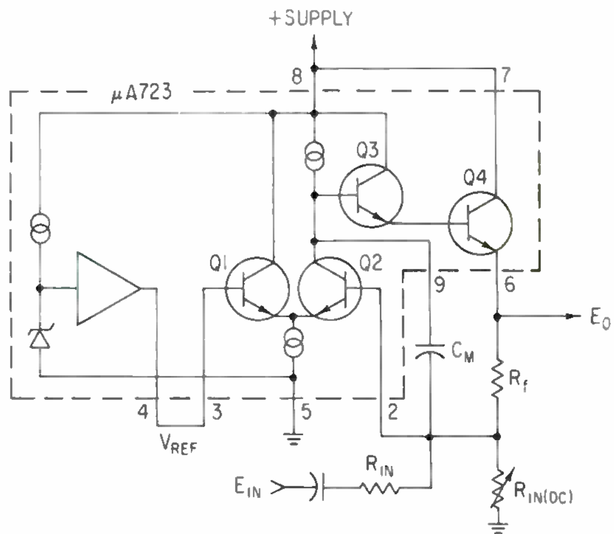

Figure 3. A μA723 connected as an audio amplifier.

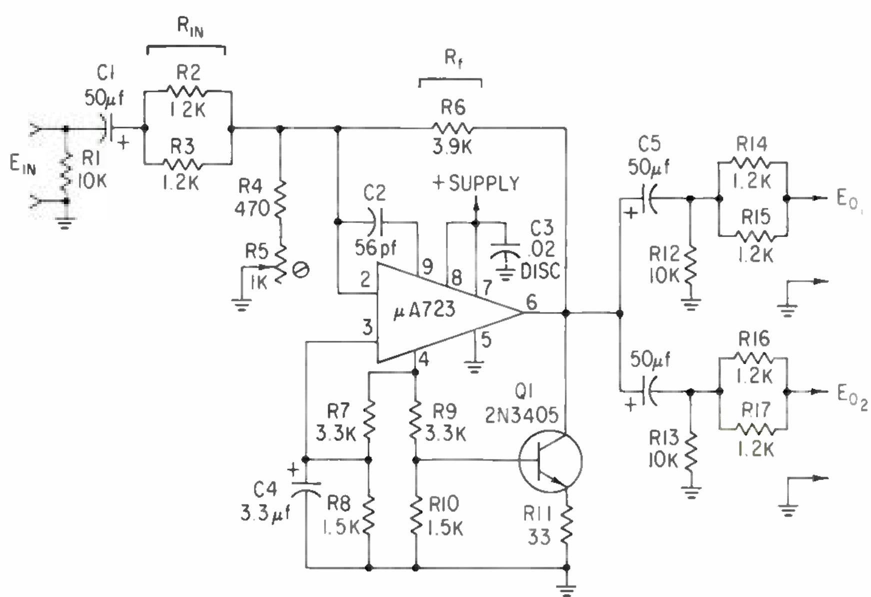

Figure 4. The complete schematic of a line amplifier using a μA723.

dBm signal into 600 ohms requires about 20 mA peak of output current-two of them twice this. A standard IC op-amp might supply it but this is beginning to push things a bit. Custom and hybrid types definitely will, but they'll also shoot down our economic goals. What we need is an IC like FIGURE 2, which is a general sketch of a circuit configuration which has the properties we desire. A simple differential pair such Q1-Q2 can develop an extremely high voltage gain for a single stage when loaded with a high-impedance collector load. This load will need to be buffered, but a Darlington pair such as Q3-Q4 will get us good load isolation and if husky enough be able to supply 50 mA or so to an external load with a low output impedance.

We can control the frequency response of this one stage op-amp very effectively by a single miller capactor (O_m) from Q2's collector to base. Q1's input will be unused in this application, or at a.c. ground. For single supply biasing we'll need some sort of voltage supply for the bases of Q1-Q2 and a current source at their emitter which can operate close to ground so we can swing large output signals. The single input supply line should feed all stages, but we would like this voltage to be non-critical so a.-c. line variations and ripple will not get through to the output.

By now it sounds as if we have backed ourselves into a corner from which only an expensive custom IC or hand-tweaked discrete kludge will extricate us. Where can we get an audio IC which will do all these things and still be within the reach of Joe Audio-Mans pocketbook? Well we can't. But don't stop here because I wouldn't have written this much without having something, would I?

Perhaps the world's best example of being able to have your cake and eat it too is a modest little TO-5 can called a μA723, which although mis-labeled as a power-supply regulator is really an audio amplifier—just as we have described. This husky little silicon gem does all the things we want for this application—it has an internal Darlington pair which can handle 150 mA of current, an internal bias reference supply which allows stable bias for the high-gain differential amp, and enough inherent rejection of input noise to allow operation from unregulated sources.

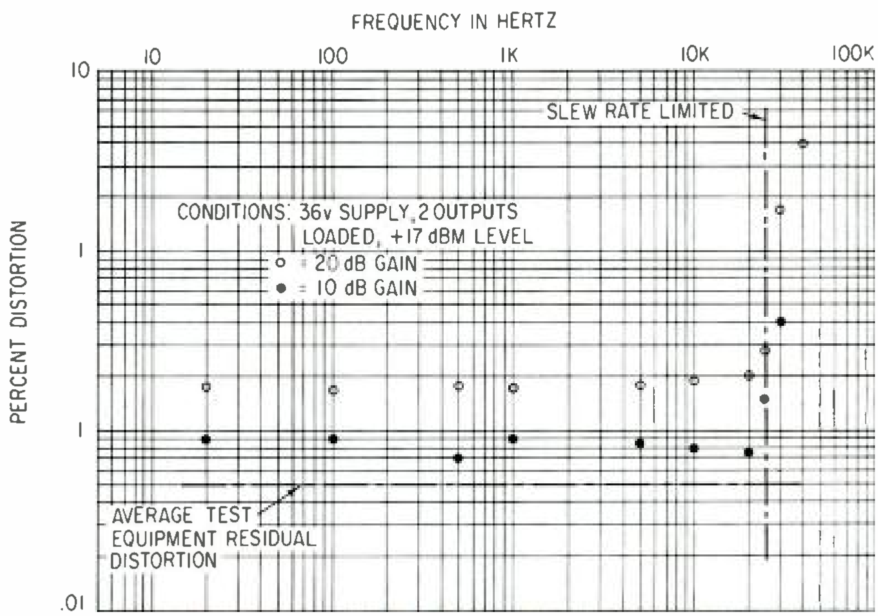

Figure 5. Total harmonic distortion at 10 dB and 20 dB gains

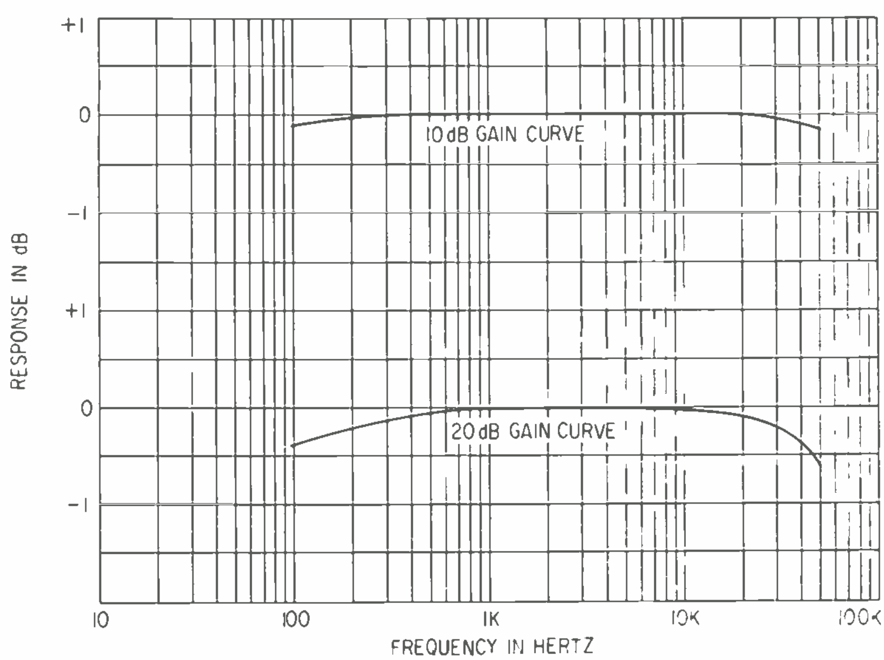

Figure 6. Open loop response with 56 pF compensation.

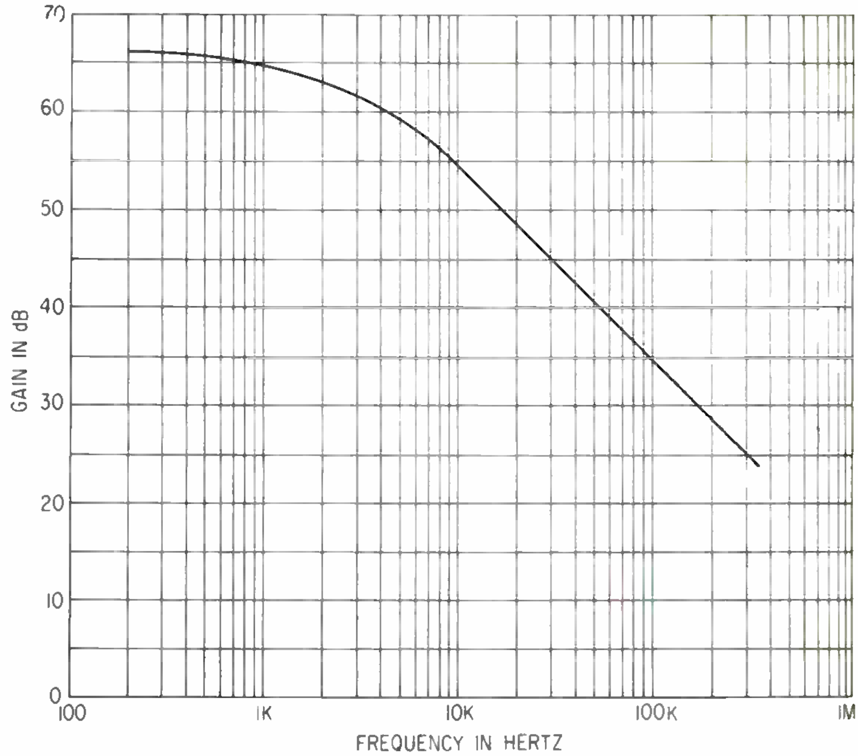

Figure 7. Relative frequency response.

The configuration and pin hookup for this IC as an audio amp is shown by FIGURE 3. The bias supply (V_{ref}) at pin 4 is applied to one side of the differential amplifier as a reference voltage (pin 3). Since Q2 is the inverting input of this operational amplifier, the input and feedback resistors are applied to this input, pin 2. Our a.-c. signal is applied via R_{in}. Since this amplifier operates all the way down to d.c., it also amplifies the voltage applied to the reference input. So for maximum a.-c. output swing we want to bias the output terminal equidistant between B+ and ground. This is done by making R_{in} (d.c.) variable so the d.-c. gain can be tweaked for symetrical clipping of positive and negative signal peaks.

Now in FIGURE 4 we turn to an actual working circuit of a line amplifier. Here we see a couple of minor modifications over FIGURE 3, inserted for reasons of improved performance.

The first is a division of the reference voltage from its nominal 7 volts down to 2.2 by R7 and R8. Since the bias of both Q1 and Q2 will ride at this d.-c. potential, this defines the lower limit of signal output swing. So by minimizing it we increase available output as we are allowing more negative signal swing. The second change is for a similar reason, to increase available output swing. This is the addition of Q1, a negative pulldown stage to linearize output swing, as Q3 and Q4 of the IC cut off during negative signal peaks. This constant-current transistor also helps lower distortion by providing a constant source impedance at all points on the output waveform.

PERFORMANCE

By now the reader is probably wondering just what do we have in the way of performance. The following data is cited as per actual measured values on the circuit of FIGURE 4. Where applicable, test conditions or circuits are shown.

Power output: Measured under the following conditions—supply equal to 36 v. both outputs loaded with 600-ohm resistive loads. P_o each output: +17 dBm before clipping. 32 Vp-p available at amplifier direct output for unmatched loads.

As it can be seen from this example the effect of the power supply voltage is to limit maximum output available. The maximum voltage that this IC can withstand is 40 volts, so the +17 dBm figure can be increased somewhat if desired. Any ripple on the supply line will be reduced by the ripple

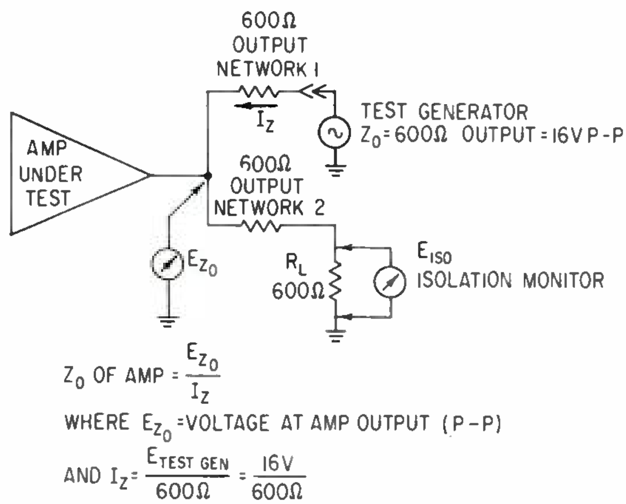

Figure 8. Signal isolation and the Z_0 test.

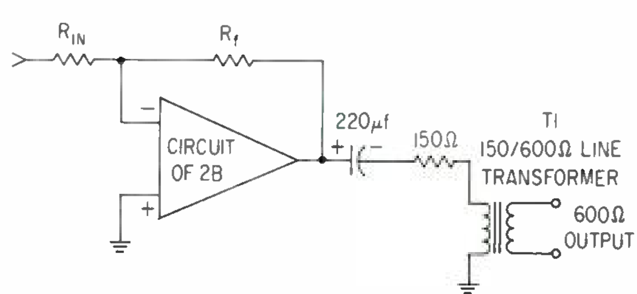

Figure 9. Alternate 600-ohm output connection for +23 dBm. You should change R11 to provide sufficient current output, and heat sinking of both the IC and Q1 is recommended. The amplifier should be labeled - - circuit of Figure 4.

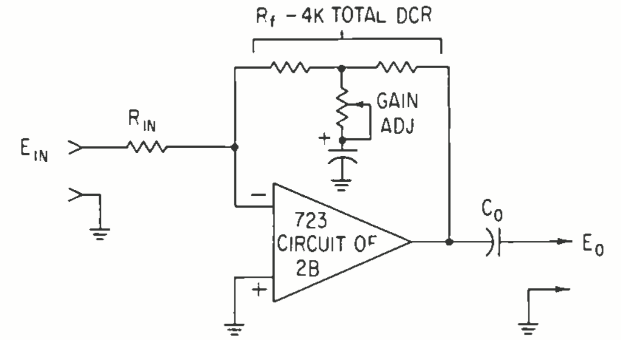

Figure 10. A configuration for adjustable gain.

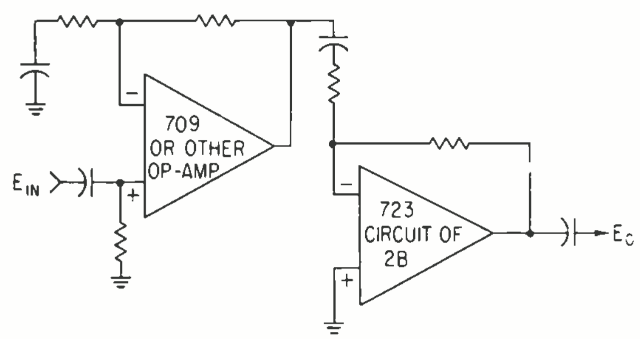

Figure 11. This configuration provides even greater gain.

rejection figure of the IC which is on the order of 70 dB. At whatever supply voltage is used, trim R5 for maximum undistorted output. Since this circuit operates class A, d.-c. power consumption is constant, in this case the drain is about 40 mA.

Distortion: Total harmonic distortion was measured under the output conditions as in test 1 and with output level adjusted to be just under the clipping level. R5 was set for symetrical clipping of positive and negative peaks by observing the distortion products on a scope and allowing the plus and minus clipping spikes to occur simultaneously.

The distortion was measured at the 10 dB gain setting and found to be under 0.1 per cent below 20 kHz (see FIGURE 5). There is a definite rise in distortion above 20 kHz due to slew rate limiting but this is felt to be of small consequence as it is outside the band of interest and only occurs at full output. The residual distortion of the oscillator-analyzer combination used averaged about 0.05 per cent across the band.

As an experiment, a 20 dB gain configuration was tried to note the effects on distortion. This is shown by the second curve; being approximately double with a proportionate rise at the high end.

Frequency Response: Once again a picture is worth a thousand words, see FIGURE 6, response at the 10 and 20 dB gains. Both are within ±0.25 dB within the audio band. A few comments are in order for those who might want to play with the response of this unit at different gains. The open loop response with a 56 pF compensation capacitor is shown by FIGURE 7. This curve has a 3 dB breakpoint around 2 kHz and a 6 dB rolloff thereafter, a typical single stage roll-off. This is in contrast to the usual IC op-amp which corners at 10 Hz. The open-loop gain at low frequencies is around 66 dB. Full power response is available up to 40 kHz.

Input-Output impedance: Not really much to say here, as it only amounts to the correct selection of resistor values. The two 1200 ohm, 5 per cent units shown give a 600-ohm match within 5 per cent or better. In both input and output positions that portion of the impedance contributed by the amplifier itself is so low as to be negligible.

Output Isolation—Line to Line: This aspect of performance is where this amplifier really shines. The method of measurement was as in FIGURE 8. A signal is pumped back in the amplifier from an external generator and the crosstalk measured at the adjacent terminal. If the amplifier was a perfect voltage source with $Z_o = 0$ no crosstalk would result of course, as no voltage could be developed across 0 ohms. The voltage that is developed is a direct measure of the Z_o of the amplifier, as the current is determined almost solely by the 600-ohm resistor.

The isolation measured between channels was >90 dB at 1 kHz. a +19 dBm signal developed a −76 dBm signal across the amplifier which is equivalent to −82 dBm signal at the adjacent output. This is close to the residual noise of the amplifier so this is really a gray area. Crosstalk is at least 95 dB down and quite possibly more.

Under most practical conditions we can safely conclude that crosstalk should not be a problem in this circuit. Crosstalk that does result will be due to coupling into the 600-ohm lines rather than through the amplifier.

Gain: As was mentioned in the initial portion, the voltage gain is a function of R_f to R_{in}. The ratio in this amplifier is 6.5 to one which gives a voltage gain of 3.25 after the 600-ohm termination, or close to the desired 10 dB. input capacitors in this circuit are chosen to be 3 dB down near 10 Hz.

Noise: Noise output is another noteworthy characteristic of this circuit. Output noise is 100 μV rms, or 93 dB below +17 dBm. Noise is affected somewhat by the bypassing on the reference supply, so if this parameter is critical a bypass is recommended across R8, such as the 3.3 μF tantalum shown.

By now the reader should be ready to take up a soldering iron and start building one of these little units. For those who might want to do just that, some general notations could be helpful.

There is really nothing about the circuit which is very critical. The only real problem might be with high frequency oscillations, and this can almost always be cured by a good high-frequency bypass on the input leads (C3 of FIGURE 4.) Use a tantalum or disc ceramic and keep leads short and direct, as if it were an r.-f. circuit. If you don't the IC will get the idea it is and take off like gangbusters.

Most of the rest of the components are straightforward. Coupling capacitors can of course be tailored to suit individual requirements. Don't try to use a single output capacitor for the two output lines and expect to get good isolation (guess who did but soon found out). The reactance of almost any reasonable capacitor will be high compared to the amplifier Z_o, especially at low frequencies and ruins isolation. The two capacitor trick gets around this one neatly, however.

The semiconductors are quite inexpensive, a factor which really makes this amplifier a good bargain. Q1 can be almost any npn which can stand the voltage and power requirements (40V,05W). The 723 is available from several manufactures now, and should be easily obtainable at modest cost.

So by now maybe the reader sees what was originally touched upon in the opening. How funny things can seem at first, but then they become quite sensible when lived with a bit. I'm now more convinced than ever that this device is really an audio IC and until now it has been disguised as a power supply. Shame! Such mundane drudgery—this little circuit deserves better. Maybe we could start a campaign to stamp out power supplies and make audio amps. . . ■

IC's The—Coming Revolution

EDWARD J. GATELY, JR.

The author makes a strong case for the present use of integrated circuits in professional audio gear, and goes on to predict their eventual dominance

SIX OR SEVEN YEARS AGO much was said and written about how transistors might be satisfactory for space capsules but they would never replace tubes in quality audio equipment. All kinds of arguments were advanced to justify the argument including the non-arguable "transistor sound". Today, little or no tube equipment is being manufactured and certainly no new designs are undergoing development. Many new equipment designs are featuring integrated circuits (IC's) and the old arguments against transistors are being heard again—only this time directed at integrated circuits.

Recent papers and articles have advanced the following anti—IC arguments.

1. Cost
2. IC's have more noise than comparable transistor circuits.
3. IC's have lower output voltage than transistor circuits.
4. IC's cannot deliver sufficient output power.

Edward J. Gately, Jr. is president of Gately Electronics, 57 West Hillcrest Avenue, Havertown, Pa. 19083

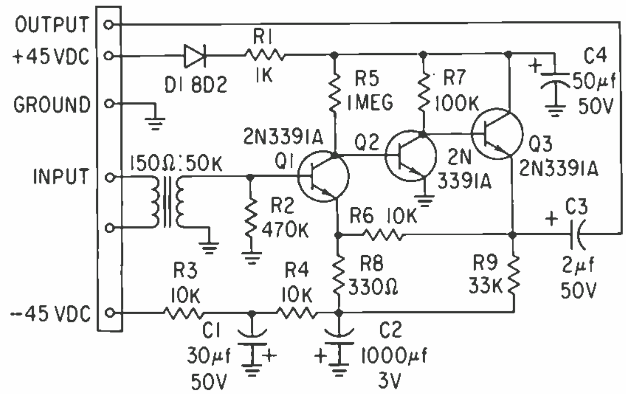

Figure 1. A mic preamp constructed with discrete components.

5. IC's are not generally available.
6. IC's are not uniform and require selection.
7. IC's require more space than comparable transistor circuits when the compensating components are considered.
8. IC circuits are difficult to service.
9. IC's have higher distortion than comparable transistor circuits.
10. IC's suffer from slew rate $\left(\frac{dv}{dt}\right)$ limitation.

FIGURES 1, 2 and 3 show typical circuits suitable for mic preamp service. One being a conventional discrete transistor amplifier, the second an op-amp version and the third an IC circuit.

FIGURES 4, 5 and 6 are line amp versions of the same three types of circuits.

CHART I presents a summary of the electrical measurements made on the preamp version. CHART 2 presents the measurements on the line amp versions.

The measurements shown indicate that all three circuits meet all the criteria of the professional circuitry. The IC is quieter but on the other hand has a slightly lower output voltage. Overall dynamic range of the three circuits are comparable except weighted slightly in the IC's favor.

Notice that both this particular op-amp circuit and the IC have to be buffered with "power" transistors to deliver sufficient level to a 600-ohm line. However the conventional transistor amplifier needs buffering in that it is a conventional voltage amplifier with a pair of emitter followers to deliver power.

The IC shown, although bearing a Motorola part number, is also available from Philco as part number FU6E7709.

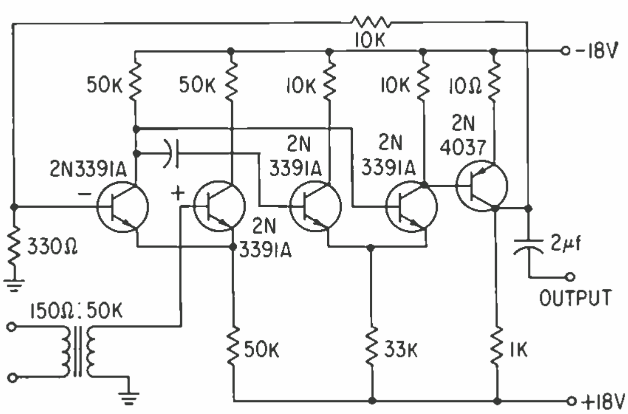

Figure 2. An opamp mic preamp.

CHART 1

Comparison of Mic Preamp Circuits

	DISCRETE	OP-AMP	IC
Noise (30hz-15khz)	−125 dBm	−124 dBm	−128 dBm
Output voltage	12 volts	12 volts	12 volts
Distortion (2)	0.2%	0.2%	0.2%
Component Cost (1)	$5.77	$4.75	$3.25

CHART 2

Comparison of Line Amp Circuits

	DISCRETE	OP-AMP	IC
Noise (30hz-15kHz)	−75 dBm	−75 dBm	−80 dBm
Output Level (4)	+28 dBm	+24 dBm	+24 dBm
Distortion (2)	0.2%	0.2%	0.2%
Component Cost (1)	$6.52	$7.06	$5.56

Notes:

1. Cost based on single quantities. Transformer prices not included. Current price MC-1709CL in single quantity $2.80.
2. Oscillator distortion 0.2%. Measurements made at output level specified at 1kHz and 10kHz into 10k-or 600-ohms load as appropriate.
3. Data reports average of measurements on three sample circuits of each design.
4. Greater output obtained for discrete line amplifier because of higher IC supply voltage. IC is limited to + and −18 volts (36 volts total).

This item is available from all major parts houses as a stock item and usually can be obtained in quantities of greater than 100 directly from stock.

Recently, to check the uniformity of IC's six samples of the Motorola MC1709CL and Philco FU6E7709 were obtained from each of three parts houses widely separated geographically. All eighteen IC's were checked for output voltage and noise in the microphone preamp shown. All units delivered a minimum of 12 volts and had equivalent noise of less than −127 dBm.

Of the three types of microphone circuits presented, the conventional discrete amplifier occupies the most space because of the required negative supply decoupling. The op-amp and IC occupy about the same amount of space. The line amp space is comparable in all three cases.

As long as the circuit designer has enough sense to use a socket under the IC (so that difficult unsoldering is not required), then IC circuit servicing is easier than circuits using discrete components. The large amounts of d.-c. feedback used in conventional circuits to insure temperature stability make it very difficult at times to determine which

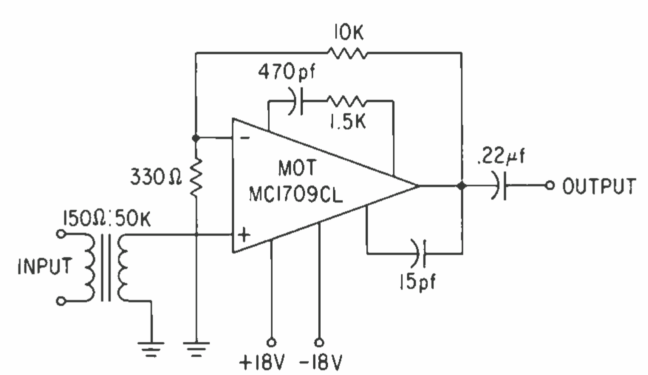

Figure 3. An integrated circuit mic preamp.

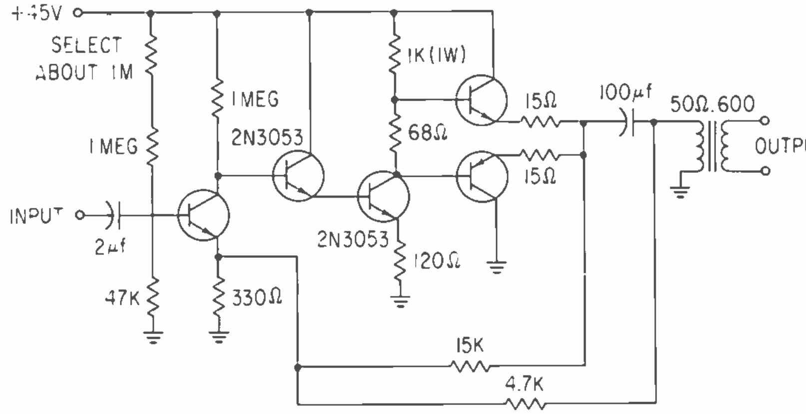

Figure 4. A line amp using discrete components.

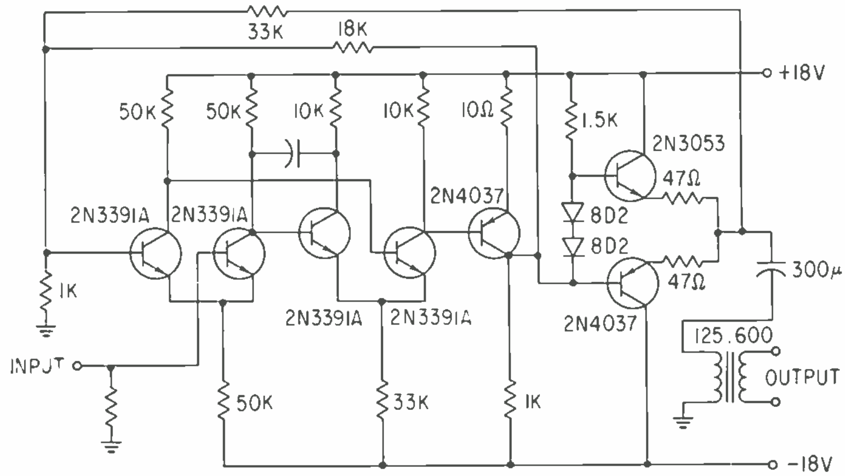

Figure 5. The same line amp as Figure 4, but using an opamp.

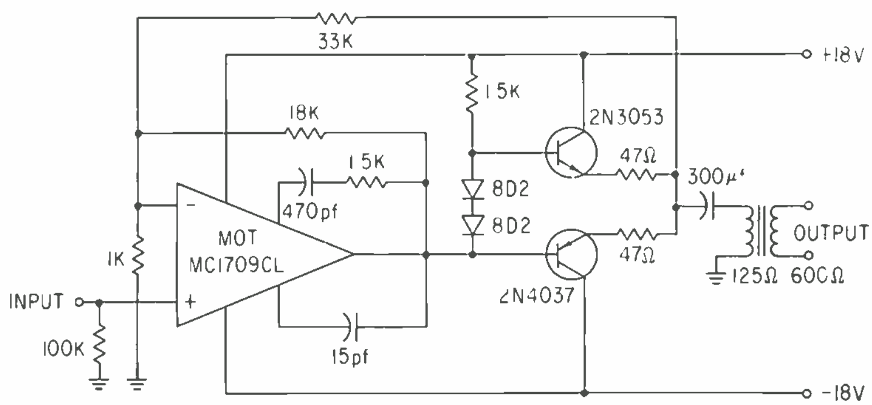

Figure 6. The line amp of Figures 4 and 5 but using the indicated integrated circuit.

of the many transistors is the faulty one. With an IC used in a circuit the whole assembly is unplugged and a new one inserted. Replacing a $2.80 IC in one minute is cheaper than replacing a $0.75 transistor if the $4.00 an hour technician takes an hour to determine which transistor to replace.

Since designers tend to select transistors for specific circuit locations it is not unusual to find six or seven types used in a single piece of equipment. This tends to complicate the spare parts problem. One IC and two transistor part numbers might be the total compliment of semiconductors in a 24 in/8 out console using the IC concept.

Distortion is hardly ever a problem in professional audio equipment obtained from a reputable manufacture, as it is so easily controlled by using adequate amounts of negative feedback. The transformers are usually the distortion limiting component. Most audio gain blocks rarely require more than 40 dB of gain. Most IC's suitable for professional circuits have open loop gains of 90 dB or more. Thus most audio integrated circuits operate with a minimum of 50 dB of negative feedback. This amount of feedback absolutely precludes any measureable distortion up to the clipping point.

Slew rate is defined as the rate of change of the IC output voltage with respect to time. In simpler terms if an IC (or other circuit for that matter) is to deliver a high output voltage at a high frequency then it must have a high slew rate. A 10-volt output at 20 kHz requires a slew rate of 2.5 volts per microsecond. When certain IC's are operated in low-gain configurations, slew-rate limitation of high-frequency signals can become a problem. This is particularly true of the new IC's which require no external compensation. However, by careful selection of compensation values and circuit design, slew rates of 5 volts per microsecond can be achieved allowing full voltage output up to 40 kHz.

With so many advantages and no real disadvantages, the swing to IC design will accelerate in the next several years. I would venture to predict that in two years more time the IC revolution will have come to pass and most if not all new designs will be around the IC format. ■

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General Specifications

- Speeds – 7½ and 3¾ ips
- Motors – 1 hyst. sync., 2 outer rotors
- Wow and Flutter – 0.12% @ 7½ ips
- Freq. Response – ±3 dB 50-15,000 Hz @ 7½ ips
- S/N Ratio – 50 dB
- Crosstalk – 48 dB

TCA-40

• ¼-track, 2-channel stereo playback, plus 4-channel stereo playback (in-line) • ¼-track, 2-channel erase and record heads for future "step-up • Automatic reverse for uninterrupted playback of conventional 2-channel tapes • Readily modified to TCA-41 or 42 • Built-in solid-state preamplifiers • Ideal for duplication master or copy deck

TCA-41 *(Illustrated)*

• ¼-track, 2-channel stereo playback, plus 4-channel stereo playback (in-line) • ¼-track, 2-channel record • Automatic reverse for uninterrupted playback of 2-channel tapes • Readily modified to future 4-channel recording capability, or TCA-42 • Solid-state playback and record preamplifiers • Off-the-tape monitoring selector

TCA-42

• ¼-track, 2-channel stereo playback, plus 4-channel stereo playback (in-line) • ¼-track 2-channel stereo record and 4-channel stereo record (in-line) • Automatic reverse for uninterrupted playback of 2-channel tapes • Total of 8 separate solid-state playback and record preamplifiers • Off-the-tape monitor selectors

HIGH FIDELITY

Let there be Quiet!

WALTER H. NELSON

The author presents practical methods by which the signal-to-noise ratio can significantly be bettered in the average studio.

HISS, OR THERMAL AGITATION NOISE, is inescapable in our amplifiers—we can only minimize it, not eliminate it. Electricity is inherently grainy.

Thermal noise arises because electrons are busy, busy dancing away their lives. The higher the temperature, the more spirited the dancing.

A moving electron constitutes a current. An electron dancing back and forth but going no place creates an *alternating* current.

As in any society, there are high-energy individuals and low-energy individuals, and some at all intermediate levels. Correspondingly, thermal alternating voltages that are present across a resistance show a very wide spectrum of frequencies, called *white noise*.

It has to get awfully cold—like near absolute zero—to discourage the electrons from dancing. Less messy ways than refrigeration are preferable for controlling thermal noise.

Our desired signal, composed of "straight" electrons moving in orderly predictable groups, must compete with the noisy individuals in the same thoroughfare, and repressive measures are effective. As the equation that is FIGURE 1 indicates, thermal noise power is proportional to resistance, so that as a practical matter, putting down the resistance represses the unwanted noise. An amplifier puts out the least noise when its input is throttled completely, that is: short-circuited. To enjoy the least disturbance we must clamp down as hard as possible on the invaders by "un-shorting" the input terminals as little as possible. Keeping the resistance low makes an effective barricade against the wildly dancing undisciplined individual electrons trying to crowd in and take over. Once inside, the noisesome individuals cannot be gotten out again, so the most critical point in the chain is the entrance gate or input terminals. With care, the first input circuit can be made to control the signal-to-noise ratio of the entire audio channel.

REDUCING HISS WITHOUT REDUCING SIGNAL

One would like to have at least 60 dB of dynamic range available when listening to good music. The human ear is a wondrous device which can appreciate much more than this range. White-noise hiss must be down at least 60 dB in

$$e_n^2 = 4KTRB$$

Figure 1. The formula for determining minimum noise.

Walter H. Nelson is with Nelson Research Laboratories in Berkeley, California.

order not to be overly intrusive. This sets a bottom limit to the signal which must be provided by the transducer (microphone, playback head, phono cartridge, etc.) of 1000 times the thermal noise. At room temperature for a band width of 25 kHz the thermal noise across 200 ohms is about 0.3 microvolt. One thousand times this amplitude is 0.3 millivolt, the least possible signal permitting a 60 dB signal-to-noise ratio. In actuality, it takes a larger signal than this because the input resistor may contribute noise on its own unless specially selected. A signal as low as 0.3 mV on 200 ohms is not unusual when microphones are used at some distance from a performing group. A spacing comparable to the dimensions of the group is often needed to obtain a tasteful balance between direct and reverberant sound and a good blend if several voices are present.

Good design of the amplifier chain can cope with this low signal. The alternative to good design is, of course, money. With a costly flock of microphones and mixers one can close-mic every individual instrument and vocalist and arrive at line-level signal amplitude without any amplification and no discernable hiss. However, the resulting signal is so dry and has so much presence that it must be processed after recording by a very artistic editor-engineer with the aid of an expensive array of multi-channel artificial reverb gimmicks to achieve an acceptable end product, what with all the unnatural squeaks, clicks and breathing sounds characteristic of close miking.

INPUT IMPEDANCE MATCHING

Stuck with a small signal for lack of this kind of money, one must rely instead on good design. (Clearly, good design requires that we minimize resistance everywhere. Following the principle of shunting down the input resistance of the first amplifier to the lowest possible value requires that we depart from the rule-of-thumb technique of matching impedances. Impedances are matched where maximum power is to be transferred. This is not the right technique when we have two kinds of signals and wish to discriminate against one of them. We want the highest *ratio* of signal to noise voltage. With microphones as the signal source, matching the impedances loses half the desired signal voltage inside the microphone, and transfers at least half of the undesired noise power to the amplifier. (According to *Electro-Voice Microphone Facts*, Volume 2 No. 1, for microphones whose impedance rises at low frequencies, such as ribbon and dynamic cardioid microphones, matching also rolls off a few dB from the low-frequency response.)

A microphone output signal nearly *doubles* when instead of matching its impedance it is operated into an impedance several times its own internal impedance. Somewhere between matched impedance and high impedance the best ratio of signal to noise is found. This is because the thermal noise at the amplifier input is shunted down more and more as the amplifier is used more nearly bridged across the mic. Going in the direction of a judicious mis-match loses noise faster than signal. Operating a microphone into an input impedance around five times its own internal impedance is a good first approximation. This ratio provides 5/6 or 83 percent of the useful microphone signal while holding down the input resistance seen by the amplifier input to a value discriminating against the thermal noise. There are refinements to this rule based on the particular characteristics of the transistor pre-amplifier in use, which are interesting to the perfectionist but too involved to treat here.

ZERO LOSS GAIN CONTROL

A fundamental fact is that *any* loss introduced into the early stages of the signal path irreparably degrades the signal to noise ratio. Variation of channel gain to accommodate different signal levels must be accomplished without throwing away any of the desired signal. This is not a contradiction in terms. Ideally, the best signal-to-noise ratio is obtained by the *non-dissipative* method of varying the negative feed-back over the first two stages, rather than by throwing away signal through a losser potentiometer such as is often found between the first two stages. The losser method of channel gain control can result in a poorer s/n ratio than is present at the microphone output. By contrast, adjusting the channel gain with an added separate control which varies the negative feedback over the first two stages provides several advantages. For one, the amplification and the signal handling capability are complementary. As the negative feedback is increased to lower the gain for higher input signal levels, the ability of the amplifier to handle these larger signals without distortion or clipping *also* increases. This precious boon results from the action of negative voltage feedback, which returns a fraction of the amplified signal to the input in phase opposition, bucking down the effective input signal without loss of signal energy into heat. By contrast, a potentiometer squanders desired signal power, converting the unused portion of the signal into heat. Drawing an analogy with a cash register—with negative feedback gain control one gets back the change, while a potentiometer keeps the change. It is the difference between amplifying too much and throwing away part of the signal, and always amplifying just enough.

A range of control of 30 or 40 dB is convenient for this separate loss-free channel gain control, which may be a recessed thumb-operated knob in line with the fader potentiometer of each channel.

Front-end negative feedback channel gain control provides other important advantages. Noise due to residual hum on the d.c. power supplied to the second stage is reduced, along with incidental noises arising in the second stage. Distortion is reduced to levels so low that the finest analyzer cannot detect any, even at levels so high that a further increase would cause clipping.

Because of this fine performance at maximum signal levels, the whole audio chain may be set up for minimum noise and maximum dynamic range at rehearsal. Faders and master gain control may be set wide open without risk of overload distortion when the channel gain is adjusted to accommodate the loudest passages, using only the separate step-less negative feedback gain control over the first two stages. This is amplifying just enough. No signal is thrown away. The softest passages are given the best chance to escape degradation by thermal noise. More attention could then be directed to quieting the ventilating system noise of the auditorium or studio, to match the performance of the electronics.

A further advantage of front-end step-less negative feedback channel gain control derives from the lowered dynamic impedance looking back into the pre-amplifier output terminals. This effectively shunts down the impedance presented to the input terminals of the next block of amplifiers in the chain, and thus helps to preserve the signal-to-noise ratio at a stage where the signal level is not yet high enough to completely swamp out the thermal noise contributed by its own input circuit.

CONTROL BANDWIDTH

Examination of the equation of FIGURE 1 indicates a second procedure that helps to control thermal noise. Noise power is proportional to band width as well as to resistance. Only the very young hear beyond 25 kHz. However, frequencies above audibility for human ears may nevertheless contribute to the audible noise by being heterodyned down into the audio band by intermodulation effects due to tape saturation at high frequencies and other nonlinearities. This is especially true if the system is ringing. A constant output signal at 30 kHz or above, with no input signal, is often observable on the oscilloscope trace of a system. This is because wide-band thermal noise is constantly exciting any and all underdamped circuits present in an audio channel. Such ringing, which occurs in response to steep wave-front transients in the desired signal as well as to the constant undesired hiss, can be traced to unterminated microphone input transformers, over-reaching record pre-emphasis networks, etc. Such hang-over effects may contribute to a loss of transparency in program material. Very wide bandwidths are useful at the two ends of an audio channel, where a microphone has better transient response if it responds to supersonics, and where a power amplifier provides tighter damping of a loud speaker when its pass band extends beyond audibility.

An internal band width of 25 kHz is sufficient, but limiting must be gentle in order to prevent ringing on steep transients containing frequencies above the nominal pass band. Roll-off of frequencies above audibility should be accomplished with single-time-constant stabilization, which limits the roll-off rate to 6 dB per octave, and the phase shift to a maximum of 90 degrees at unity gain. This roll-off rate is sufficiently rapid to dispose of the AM broadcast stations heard through some super bandwidth amplifiers. In locations near radio stations where strong radio-frequency fields exist, foreign currents flowing in early amplifier stages may increase the hiss level even though no program is heard. Another advantage of technique number one—keeping impedance levels low—derives from the fact that relatively large capacitors can be used to by-pass unwanted r.f. signals, along with very small diameter shielded cables, without interfering with the 25kHz internal bandwidth.

SUMMARY

Thermal agitation noise is an inherent property of electricity and can not be eliminated, only minimized. It is proportional to resistance, bandwidth and temperature. Resistance and bandwidth can be minimized to reduce noise without reducing the desired signal, but minimizing temperature is impractical.

Pay the most attention to the low-signal-level first-input-circuit, which can be made to control the s/n ratio of the entire system. Avoid attenuator coupling networks between blocks of amplifiers because they force the use of higher gain which then amplifies hiss more than signal. Effect a judicious mis-match between microphone output impedance and mic pre-amplifier input impedance, in order to favor the desired signal over the undesired thermal agitation noise. Use loss-less channel gain control in addition to standard faders and master gain control, to obtain maximum possible s/n ratio along with capability of handling the large signals from high-output microphones which provide maximum dynamic range. Restrict the internal band width to 25 kHz, but with gentle roll-off, to prevent ringing on unrestricted input signals. Avoid all underdamped circuits, to escape ringing excited by white noise and input transients. ■

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SONY SUPERSCOPE

Circle 39 on Reader Service Card

39th AES Convention and Exhibition

QUICK SUMMARY

Hotel New Yorker, 34th Street and 8th Avenue
New York City

SUNDAY, October 11—Welcoming Cocktail Party
6:00 to 7:30 P.M. Terrace Room

REGISTRATION

Mezzanine

Monday, Oct. 12th—8:00 A.M. to 8:00 P.M.
Tuesday, Oct. 13th—8:30 A.M. to 8:00 P.M.
Wednesday, Oct. 14th—9:00 A.M. to 5:00 P.M.
Thursday, Oct. 15th—9:00 A.M. to 5:00 P.M.

EXHIBIT HOURS

Mezzanine, 3rd and 5th Floors

Monday and Tuesday, October 12 and 13 — 1:00 to 9:00 P.M.
Wednesday and Thursday, October 14 and 15 — 11:00 A.M. to 5:00 P.M.

TECHNICAL SESSIONS

Monday, Oct. 12

9:00 A.M.—Annual Business Meeting
9:30 A.M.—Transducers
2:00 P.M.—Electronic Music
2:00 P.M.—Standardization of Stethoscopes and Audio in Medicine—1970
7:30 P.M.—Four-Channel Recording and Reproducing Techniques
7:30 P.M.—Workshop on Stethoscopes—Hartford Rm. (See Bulletin Board)

Tuesday, October 13

9:00 A.M.—Disc Recording and Reproduction I
2:00 P.M.—Disc Recording and Reproduction II
2:30 P.M.—Broadcasting
7:00 P.M.—Studio Recording Techniques Today

Wednesday, Oct. 14

9:30 A.M.—Magnetic Recording and Reproduction
2:00 P.M.—Sound Reinforcement and Architectural Acoustics
2:30 P.M.—Audio Instrumentation and Measurements
7:00 P.M.—Social Hour—New Orleans Room
8:00 P.M.—Awards Banquet—Terrace Room

Thursday, Oct. 15

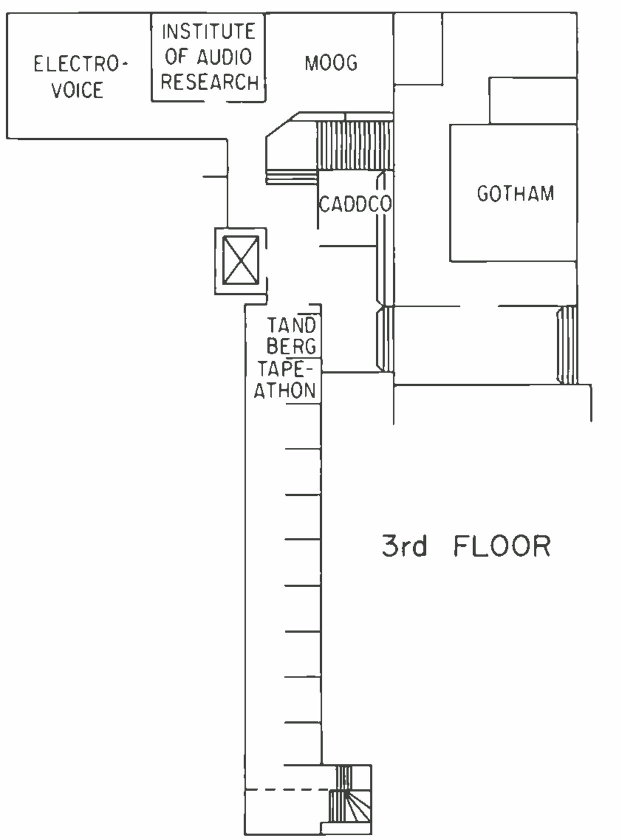
9:30 A.M.—Audio Transmission Systems: Theory, Standards, and Practice
2:00 P.M.—Amplifiers and Audio Circuitry

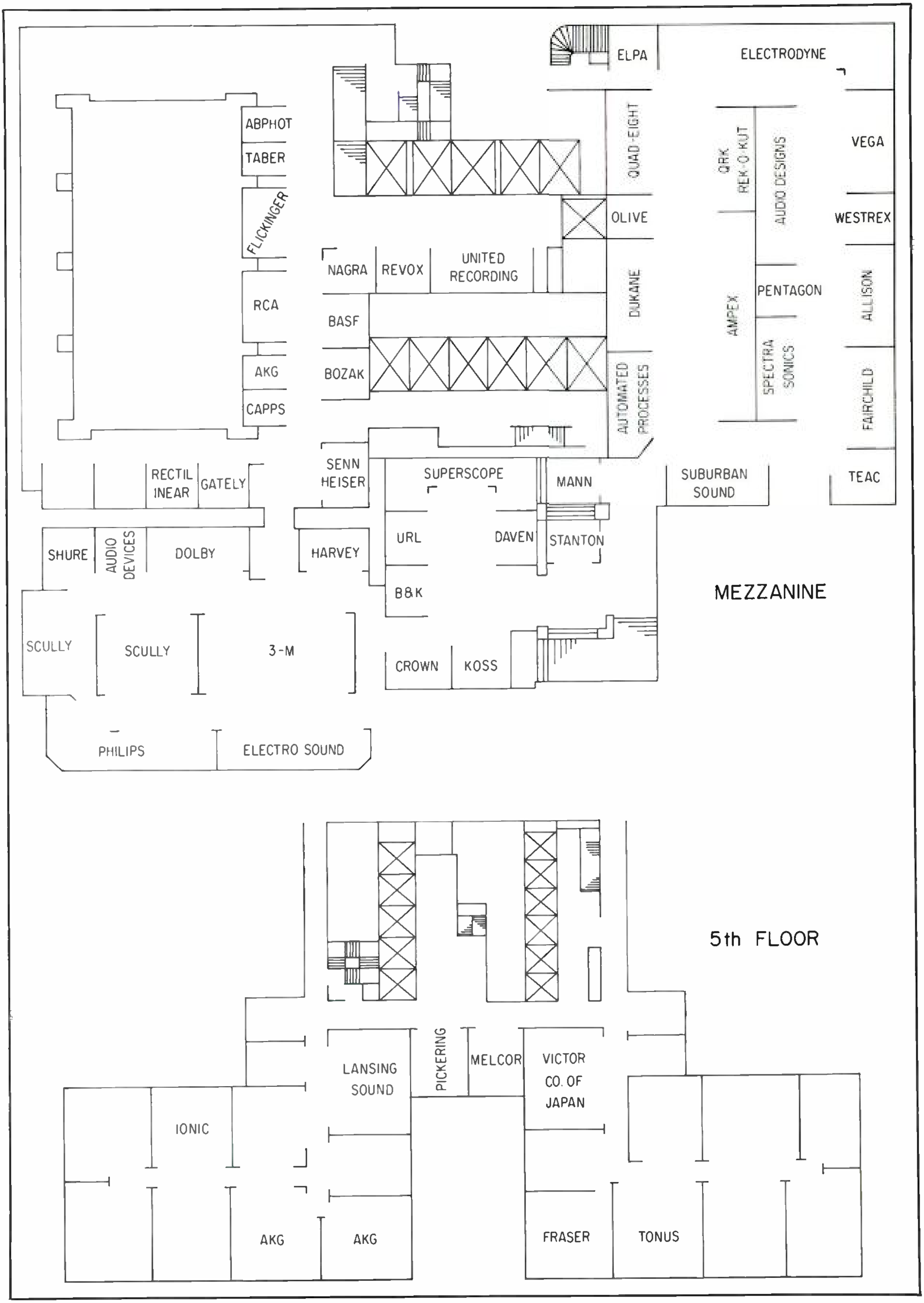
BANQUET AND SOCIAL HOUR

7:00 P.M.—Social Hour—New Orleans Room
8:00 P.M.—Banquet—Terrace Room

LADIES PROGRAM

9:30 A.M.—Monday, Tuesday and Wednesday—
Coffee Hour—AES Suite
Monday, October 12, 9:30 A.M.
Terrace Room

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THE PAPERS

TRANSDUCERS

Chairman: EDWARD R. HANSON,
North American Philips Corp., New York, N. Y.

Some Design Considerations For Electrostatic Headphones—John J. Bubbers, Stanton Magnetics, Plainview, N. Y.

A Three-Way Columnar Loudspeaker for Reinforcement of the Performing Arts—Alan P. Smith, Consultant in Electroacoustics, Maitland, Florida

The Sound Field in Home Listening Rooms—Roy F. Allison and Robert Berkovitz, Acoustic Research, Inc., Cambridge, Massachusetts

The Colinear Array—A Two-Way Loudspeaker System for Sound Reinforcement—G. L. Augspurger, James B. Lansing Sound, Inc., Los Angeles, California

Active and Passive Filters as Loudspeaker Crossover Networks—J. Robert Ashley and Allan Kaminsky, University of Colorado, Colorado Springs, Colorado

Transducer, Preamplification and Powering Technology of Transistorized Condenser Microphones—B. Weingartner, AKG Microphones, Vienna, Austria

Miniature Headband Mounted Dynamic Microphone for Professional Applications—Alan R. Watson, Electro-Voice, Buchanan, Michigan

Monday, October 12, 2:00 P.M.
Terrace Room

ELECTRONIC MUSIC

Chairman: ROSS BROWN,
CBS Musical Instruments, Fullerton, California

A New Easy-Play Keyboard Instrument—Larry Chmiel and Charles Tennes, Hammond Organ Company, Chicago, Illinois

Compositional Considerations in Electronic Music—Hubert S. Howe, Professor of Music, Queens College, New York

The ARP Synthesizer—A New Instrument for Musical Composition and Performance—David Friend, Tonus Inc., Newton Highlands, Massachusetts

A Low Cost Educational Music Synthesizer Concept—Ralph W. Burhans, Ohio University, Department of Electrical Engineering, Athens, Ohio

The Putney: A New Generation of Synthesizers——Alfred Mayer, Ionic Industries Incorporated, Morristown, New Jersey

The Versatile Electro Comp—Jeff Murray, Dale Blake Fred Locke, Norm Milliard, Electronic Music Laboratories, Inc., Hartford, Connecticut

Compact Performance Synthesizer Design Considerations—William R. Hemseth and Robert A. Moog, R. A. Moog, Inc., Trumansburg, N. Y.

Monday, October 12, 2:00 P.M.
New Orleans Room

STANDARDIZATION OF STETHOSCOPES AND AUDIO IN MEDICINE—1970

Chairman: PHILIP KANTROWITZ,
Consultant on Bioengineering, New York, N. Y.

A Special Report From the Stethoscope Committee*—How Audio Engineers can help improve Stethoscopy

Coordinators: DALE GROOM, M.D., Professor of Medicine, University of Oklahoma, School of Medicine
PAUL Y. ERTEL, M.D. Associate Professor of Pediatrics, Ohio State University

Part I. An historic perspective; the impact of Laennec's stethoscope from 1816 to today.

Part II. Barriers to understand the stethoscope acoustically: how signals are generated within the heart and reach the ear; the impedance of chest wall and of the ear; and what we need to know about acoustic pathway.

Part III. The state of the art: how stethoscopes and stethoscope components can distort sound; special stethoscopes for special purposes; facts and fiction.

Part IV. The clinical significance of cardiac auscultation: what engineers should know about acoustic signals from the heart; the relationships of cardiac sounds to hemodynamics and cardiac function; how stethoscopes can make earlier diagnoses, and last but most important—how the audio engineer can help improve stethoscopy.

An open workshop follows at 7:30 in the evening. Everyone is invited to attend this significant discussion. See Bulletin Boards for location.

*Sub-Committee of the AES Standards Committee

Auditory Aids and Substitute Devices—George W. Fellendorf, Alexander Graham Bell Association for the Deaf, Washington, D. C.

A Typical Day in an Intensive Care Unit—Eugene P. Harter, Mennen-Greatbatch Electronics, Inc., Clarence, N. Y.

Understanding the Inner Ear Through Models: Studies in Facts and Artifacts—Martin B. Lesser and David A. Berkley, Bell Telephone Laboratories, Whippany and Holmdel, New Jersey

Stroke Volume and Cardiac Output by Echocardiography—Benedict Kingsley, Hahnemann Medical College & Hospital, Philadelphia, Pennsylvania

Development of a Biocompatible Natural Rubber for Implantable Artificial Organs or Instrumentation—K von Dally, Y. Imai, D. Peabody, and Y. Nose, Artificial Organs Research Laboratory, The Cleveland Clinic Foundation, Cleveland, Ohio

An Application of Differential Ultrasonic Spectroscopy to Red Blood Cell Measurements—Marc Mangot, New York University, Department of Electrical Engineering, New York, N. Y.

Monday, October 12, 7:30 P.M.
Terrace Room

FOUR-CHANNEL RECORDING AND REPRODUCING TECHNIQUES

Chairman: JAMES H. CUNNINGHAM, 8-Track Recording Company, Chicago, Illinois

One Plus One Equals Four—Peter W. Tappen, Bolt Beranek and Newman Inc., Downers Grove, Illinois

The Effect of Microphone and Loudspeaker Directional Characteristics Upon Recreating Acoustic Fields—Edward M. Long, Ampex Corporation, Elk Grove Village, Illinois

The Simulation of Moving Sound Sources—John M. Chowning, Stanford University, Department of Music, Stanford, California

FOUR CHANNELS AND COMPATIBILITY—Peter Scheiber, Audiodata Co., Peekskill, N. Y.

On the Processing of Two and Three-Channel Program Material for Four-Channel Playback—John M. Eargle, Mercury Record Productions, New York, N. Y.

Experiments in Four Channel Recording Techniques—John M. Woram, RCA Records, New York, N. Y.

Tri-Wave Stereo Acoustics—John E. Volkmann, RCA Laboratories, Princeton, New Jersey

Tuesday, October 13, 9:00 to 11:00 A.M.
Terrace Room

DISC RECORDING AND REPRODUCTION I

Chairman: JAMES H. KOGEN, Shure Brothers, Incorporated Evanston, Illinois

Audio Developments in Europe—John C. G. Gilbert, Northern Polytechnic, London, England

Progressive Record Wear—An Optical Study—Ian Hamilton and James C. Lewis, Imperial College, Mechanical Engineering Dept. London, England

Analysis of Crosstalk in Stereo Discs—Bernhard W. Jakobs, Shure Brothers, Incorporated, Evanston, Illinois

A New Profile for LP Records—Warren Rex Isom, RCA Records, Indianapolis, Indiana

Phase Shift Characteristics of Record Cutters and Pickups—B. B. Bauer, D. Gravereaux, A Gust, CBS Laboratories, Stamford, Connecticut

11:00 A.M. to 12:30 P.M.

A Mechanical Disc Recording and Reproducing System with High Storage Density and Information Flux—Horst Redlich, TELDEC, Hans-Joachim Klemp, TELDEC, and Gerhard Dickopp, AEG-Telefunken

Tuesday, October 13, 2:00 P.M.
Terrace Room

DISC RECORDING AND REPRODUCTION II

Chairman: JAMES H. KOGEN, Shure Brothers, Incorporated Evanston, Illinois

British Contributions to Audio During the Past Fifty Years—Percy Wilson, Percy Wilson and Partners, Oxford, England

New Criteria for Stereo Disc Tracking—Lawrence Shaper, Empire Scientific, Garden City, N. Y.

An Electronic Speed Controlled Turntable for Broadcasting Application—Frank H. Hirsch, Thorens-Franz AG, Wettingen, Switzerland

A Discrete Four-Channel Disc and its Reproducing System—T. Inoue, N. Takahashi and I. Owaki, Victor Company of Japan, Limited (JVC America, Inc.), Yokohama, Japan

A Theory of Scanning Loss of Phonograph Play-Back—James V. White, Division of Engineering and Applied Physics, Harvard University, Cambridge, Massachusetts

Interaction Between Tracing and Deformation Errors—Duane H. Cooper, University of Illinois, Urbana, Illinois

Clear Sound Records Applied with PTS System—Motokazu Ohkawa, Mamoru Kuriyagawa and Shinichi Makino, Toshiba Research & Development Center, Kawasaki, Japan

Tuesday, October 13, 2:30 P.M.
New Orleans Room

BROADCASTING

Chairman: A. C. ANGUS, Daven Division, McGraw Edison, Manchester, New Hampshire

Two Portable Reproducing Consoles for Sound Effects Operations—John R. Gable, Jr., American Broadcasting Co., New York, N. Y.

The Design of a New Modular Audio Console for Broadcasting and Recording—W. F. Hanway, RCA, Meadow Lands, Pennsylvania

Applications of Audio Signal Processing Devices—Emil L. Torick, CBS Laboratories, Stamford, Connecticut

Comparison of Crosstalk Characteristics of Shielded Wiring and Printed Circuit Track—J. A. Wissner, RCA, Meadow Lands, Pennsylvania

Possible Methods for FM Broadcast Transmission of Four Channel Stereo Signals—Leonard Feldman, S.C.A. Services Co., Inc., Great Neck, N. Y. and William S. Halstead, RTV International, Inc., New York, N. Y.

An Advanced Volume Level Indicator—William P. Brandt, Altec-Lansing, a division of LTV Ling-Altec, Inc., Anaheim, California

Tuesday, October 13, 7:00 P.M.
Terrace Room

A SPECIAL EVENING MODERN RECORDING STUDIO TECHNIQUES

Chairman: WILLIAM E. WINDSOR, *D. B. Audio Corp., New York, N. Y.*

A LIVE DEMONSTRATION

Engineer: JOHN WORAM, RCA Records, New York, N. Y.

Producers: STEVE SCHWARTZ, *RCA Records, New York, N. Y.*
MAX WILCOX, *RCA Records, New York, N. Y.*

Panelists: DAVID GREENE, *A & R Recording, New York, N. Y.*
ROY HALLE, *Columbia Records, New York, N. Y.*

The New York Section of the Audio Engineering Society presents this applications seminar as an educational service to all members and guests of the Society. All those professionally engaged, active or interested in professional recording techniques are invited to attend.

This applications seminar will be a live and pre-recorded demonstration of multi-track recording and mixdown techniques, showing exactly how a modern recording is made.

Excerpts from an actual recording session will be used, and additional guitar, and Moog Synthesizer tracks will be recorded during the session. The orchestra used in the recordings includes a seven man rhythm section and a seven man brass section. Horns and drums are recorded in the classical two-channel stereo technique, while the other musicians are each assigned their own tracks on the multi-track recorder. The record producers will explain what they were trying to achieve in the recording, and additional slides will show miking techniques used in the session.

After the demonstration, the engineering panelists will discuss other recording techniques as used in today's studios and then open the Session for a question and answer period.

There will be no fee for registration for this Session.

Wednesday, October 14, 9:30 A.M.
Terrace Room

MAGNETIC RECORDING AND REPRODUCTION

Chairman: STEWART L. SMITH,
GRT Corporation, Sunnyvale, California

A Dropout Detector for Testing Multi-Channel Tape for Re-Use—Michael McLean, Motown Record Corporation, Detroit, Michigan

Tape Noise in Audio Recording—Eric D. Daniel, Memorex Corporation, Santa Clara, California

Quality of Low Speed Tape Copies—Keith O. Johnson, Gauss Electrophysics, Los Angeles, California

Computerized Re-mastering in the Manufacture of Slow Speed Tape Records—Richard Erickson and Walter Goldsmith, Certron Corp., Anaheim, California

Dolbyized Duplicating, its Effects on the Pre-Recorded Cassette—David Sarser, Allison Audio Products, Inc., Hauppauge, N. Y.

Inspection and Evaluation of Audio Recording Tape—James B. Wood, GRT Corporation, Sunnyvale, California

A Plastic Pressure Roller for Stereo-8 Cartridges—Warren Rex Isom, RCA Records, Indianapolis, Indiana

Wednesday, October 14, 2:00 P.M.
Terrace Room

SOUND REINFORCEMENT AND ARCHITECTURAL ACOUSTICS

Chairman: DAVID L. KLEPPER,
Bolt Beranek and Newman Inc., New York, N. Y.

Innovations in Studio Design and Recording in the Victor Record Studios—Yuma Shiraishi, Kiyoshi Okumura and Masahiro Fujimoto, Victor Company of Japan, Ltd. (JVC America, Inc.), Yokohama, Japan

Componentized Architectural Acoustics — Curt Knoppel, Intertex Corporation, Wauwatosa, Wisconsin

Isolation Recording, a Producer's View—John Rhys, Sonad Electron Corporation, Wauwatosa, Wisconsin

RCA Variable Acoustics Studios—John E. Volkmann RCA Laboratories, Princeton, New Jersey

Auditorium Acoustics Simulator; Form and Uses—Thomas R. Horrall, Bolt Beranek & Newman Inc., Cambridge, Massachusetts

Masking Noise Systems and Open and Closed Spaces—Ranger Farrell, Ranger Farrell and Associates, Consultants, Irvington-on-Hudson, N. Y.

A Sound System for Amphibious Assault Training—Allan P. Smith, U.S. Naval Training Device Center, Orlando, Florida

Sound System Design for St. Mary's Cathedral, San Francisco, California—Charles J. Catania, San Rafael, California

Wednesday, October 14, 2:30 P.M.
New Orleans Room

AUDIO INSTRUMENTATION AND MEASUREMENTS

Chairman: STEPHEN F. TEMMER,
Gotham Audio Corporation, New York, N.Y.

An Automatic Highway Noise Monitor—Richard G. Allen, Thomas P. Owen and Emil L. Torick, CBS Laboratories, Stamford, Connecticut

Accurate Measurements Using an Infrasonic Acoustic Comparator—Robert O. Fehr, Fehr & Fiske, Inc., Westport, Connecticut

Computerized Production of Reproduce Alignment Tapes—Richard W. Erickson, Certron Corporation, Anaheim, California

Simple and Complex Test Signals for Music Reproduction Systems—Thomas A. Saponas, Randolph C. Matson and J. Robert Ashley, University of Colorado, Colorado Springs, Colorado

Impulse Measurement Techniques for Quality Determination in Audio Equipment — Alfred Schaumberger, Audio Consultant to Georg Neumann GmbH Electroacustic, Berlin, West Germany (Translated and presented by Stephen F. Temmer, Gotham Audio Corporation, New York, N.Y.)

Thursday, October 15, 9:30 A.M.
Terrace Room

AUDIO TRANSMISSION SYSTEMS: THEORY, STANDARDS, AND PRACTICE

Chairman: JOHN G. McKNIGHT,
Ampex Corporation, Redwood City, California

Panel Members: A. PIERCE EVANS,
Audio-Video Section, Engineering and Development Department, CBS Television Network, New York, N.Y.

GEORGE MALING,
IBM Acoustics Laboratory, Poughkeepsie, N.Y.
DOUGLAS SMITH,
Electronic Development Department, Shure Brothers, Evanston, Illinois

A series of five separate panel discussions of various aspects of the subject, with introductions by the panel, and opportunity for audience participation.

1. The need for simplicity and flexibility in audio transmission system design and operation.
2. Some concepts and terminology: "matching"; levels and decibels; available power; the various kinds of "gains".
3. Design techniques: for the system (putting together a "power matched" system, a "voltage matched" system, and a "mixed system); for the elements (designing for these different kinds of systems).
4. Measurement techniques: equipment used; the test setup; the measurement procedure; conversions between the different kinds of "gain"; presenting the results as specifications.
5. Standards: What standards exist, and what do they say? Should the standards be changed; and if so, how?

Thursday, October 15, 2:00 P.M.
Terrace Room

AMPLIFIERS AND AUDIO CIRCUITRY

Chairman: FRED L. MERGNER,
Fisher Radio Company, Long Island City, N.Y.

Investigation of Various Forms of Distortion inherent in Transistor Amplifiers—Shinichi Ohashi, Hitachi, Ltd., Tokyo, Japan

A Dynamic Noise Filter—Richard S. Burwen, Consulting Electronics Engineer, Lexington, Massachusetts

A New Modular Console Building Block Concept Using Integrated Circuits—John H. Buffington, Gately Electronics, Havertown, Pennsylvania.

A Wide Dynamic Range Limiter and Program Conditioner—David E. Blackmer and Saul A. Walker, Automated Processes, Inc., Farmingdale, N.Y.

Functional Protection of High-Power Amplifiers—Max Scholfield and Gerald Stanley, Crown International, Elkhart, Indiana

A Noise Reduction System for Consumer Tape Applications—Ray M. Dolby, Dolby Laboratories Inc., London, England

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People, Places, Happenings

●Two promotions at **Pickering and Company, Inc. George P. Petetin** will assume the position of OEM sales manager for both the Pickering Audio and Pickering Measurement & Controls divisions. Prior to this appointment he was sales manager for the **Stanton Magnetics, Inc.** division. Except for a period with **Reeves Soundcraft,** he has been with Pickering since 1950. In making the announcement, Pickering director of marketing **Dan Collins** stated that this is part of a restructuring designed to provide greater concentration in some of the more important marketing areas.

Replacing Mr. Petetin as sales manager of Stanton Magnetics is **Joseph S. Woodstock** who had handled Pickerings dealer sales. Stanton Magnetics supplies professional products for the recording and broadcast industries, while Pickering manufactures for the OEM and consumer fields.

●**Custom Audio** of Des Plaines, Illinois has instituted a complete script-to-sound service. They can now begin with a basic idea or film script, construct a finished script, supply narrator, recording, visuals, music, duplication, packaging, labeling, and mailing.

●**Lawrence LeKashman,** president of **Electro-Voice, Inc.** has been elected a corporate vice-president of the parent company, **Gulton Industries, Inc.** His new responsibilities will include the management and direction of all phases of Gulton's electro-acoustic group. This consists of the company's E-V subsidiary with facilities in Buchanan and Niles, Michigan; Newport and Sevierville, Tennessee; and the newly-acquired Game Industries in Freeport, N.Y. Mr. LeKashman joined E-V in 1957 as vice-president marketing. On December 9, 1968 he was elevated to the presidency.

●A new chair in music has been created at the **University of Surrey** in England. As part of the B. Music degree to be offered, a specialized course in electronics given in conjunction with the physics department, will produce graduates that are fully competent in both the technical and artistic aspects of music reproduction to meet the increasing needs of the broadcast and recording industries. The **Tonmeister** course, as it is called, satisfies the European requirements for recording engineers to be both technically and musically qualified. The European engineer does all that American recording engineers are required to do, and additionally is fully responsible to realize for the listener the intentions of the composer and interpreter.

●**General Radio Company** and **Grason-Stadler Company,** both of West Concord, Mass., have announced their agreement for the merging of the two firms. Grason-Stadler, under its existing management, will operate as a wholly-owned GR subsidiary. Grason-Stadler Company has been in business for twenty years and manufactures precision audiometers and instruments for experimentation in psychoacoustics and the life sciences generally.

●Jacksonville, Illinois area listeners now have a new automated stereo station that operates automatically from 5:45 to midnight. **WEAI-FM** automated and has increased the variety of its programming as a result. According to operations manager **Wayne Edwards, Arnie** (as the machine has been nick-named) is automated but not computerized. The system is programmed so that there are at least 15 minute intervals between each advertiser's commercial. And if important news or weather bulletins occur, Arnie can be stopped for live broadcast, after which programming will pick up smoothly.

●An announcement from **DuKane Corporation** tells of the appointment of **Clarence Kaebnick** to the newly-created position of director of manufacturing, as well as the material control operation, for all four divisions of DuKane. The creation of this position was necessitated by the planned expansion of these divisions which are: communications systems, audio-visual, special products, and ultrasonics. Mr. Kaebnick comes to DuKane from **Ampex** where he was a vice-president manufacturing.

●The **Eleventh Acoustical Training School,** conducted by **Michael J. Kodras** and **Robert Lindahl** will be held at the Dearborn Inn, Dearborn, Michigan on November 9 and 10, 1970. The two-day session will be devoted to general principles of architectural acoustics, sound transmission loss, acoustical correction of rooms, industrial noise, and noise control of air conditioning, heating, and ventilating equipment. The course is designed for engineers, architechs, etc., and has, in years past, been over-subscribed as the mimber or registrants is limited.

QRK *Announces a Breakthrough*

QUALITY CONSOLES AT REALISTIC PRICES!

QRK-5
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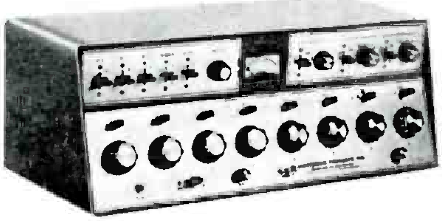

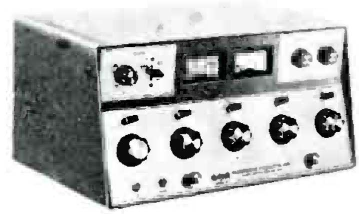

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QRK-8/8S — 8 Channel Console — QRK offers a professional console with Altec faders; plug-in modules (3) pre-amplifiers; built-in power supply; 10 watt monitor amplifiers; independent audition and program channels; muting relays; cue amplifiers; built-in speaker; substantial capacity and ultimate access.

QRK-5 — 5 CHANNEL MONO $995
QRK-5S — 5 CHANNEL STEREO . . $1595

QRK-5/5S — 5 Channel Console — Both mono and stereo units incorporate Altec attenuators with cue switches in every fader, 10 watt monitoring amplifiers, plug-in modules, muting relays, and self-contained power supply. The stereo unit, QRK-5S contains independent audition and program channels as well as a cue amplifier. Both consoles have substantial capacity and total access.

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716 Jersey Ave., Gloucester City, N. J. 08030
Phone: (609) 456-1716

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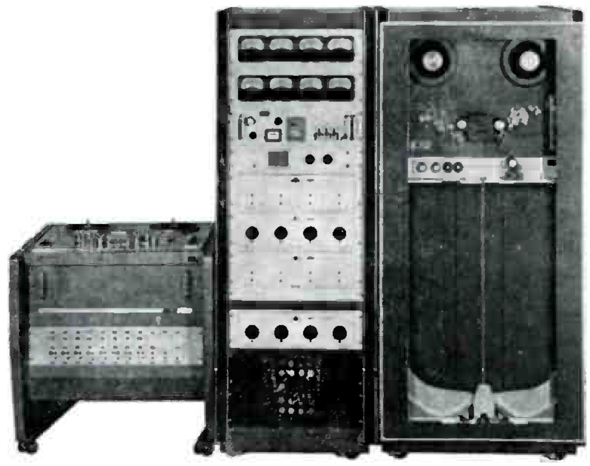
Circle 11 on Reader Service Card

32:1 A new generation of endless loop tape duplicators has arrived!

It's the Otari generation. State-of-the-art, field tested and proved in production at world-leading duplicating centers. With 240 ips tape speeds that duplicate a 30-minute cartridge program in just 15 seconds. With superior performance specs that ensure consistently high sound quality and highest yield rates. With ruggedness and dependability built in to a degree the U.S. tape industry hasn't seen before. The Otari generation is here and now: available for demonstration and backed by factory-trained engineering, service and parts. If you're producing sounds worth listening to, you'd better listen to Otari. For details and/or a demonstration, write or call Jack Ames at Otari of America, Ltd., 8295 South La Cienega Boulevard, Inglewood, California 90301 / Telephone: (213) 678-1442

OTARI

A NAME WORTH REPEATING



Up to 20 Slaves Complete Otari duplicating system consists of Master Reproducer with vertical endless loop bin, Logic Controller Console, and up to twenty Slave units for high volume production. Three systems available: 8-track, Cassette or Compatible System with interchangeable heads and guides.