dB

THE SOUND ENGINEERING MAGAZINE

MAY 1971 75¢

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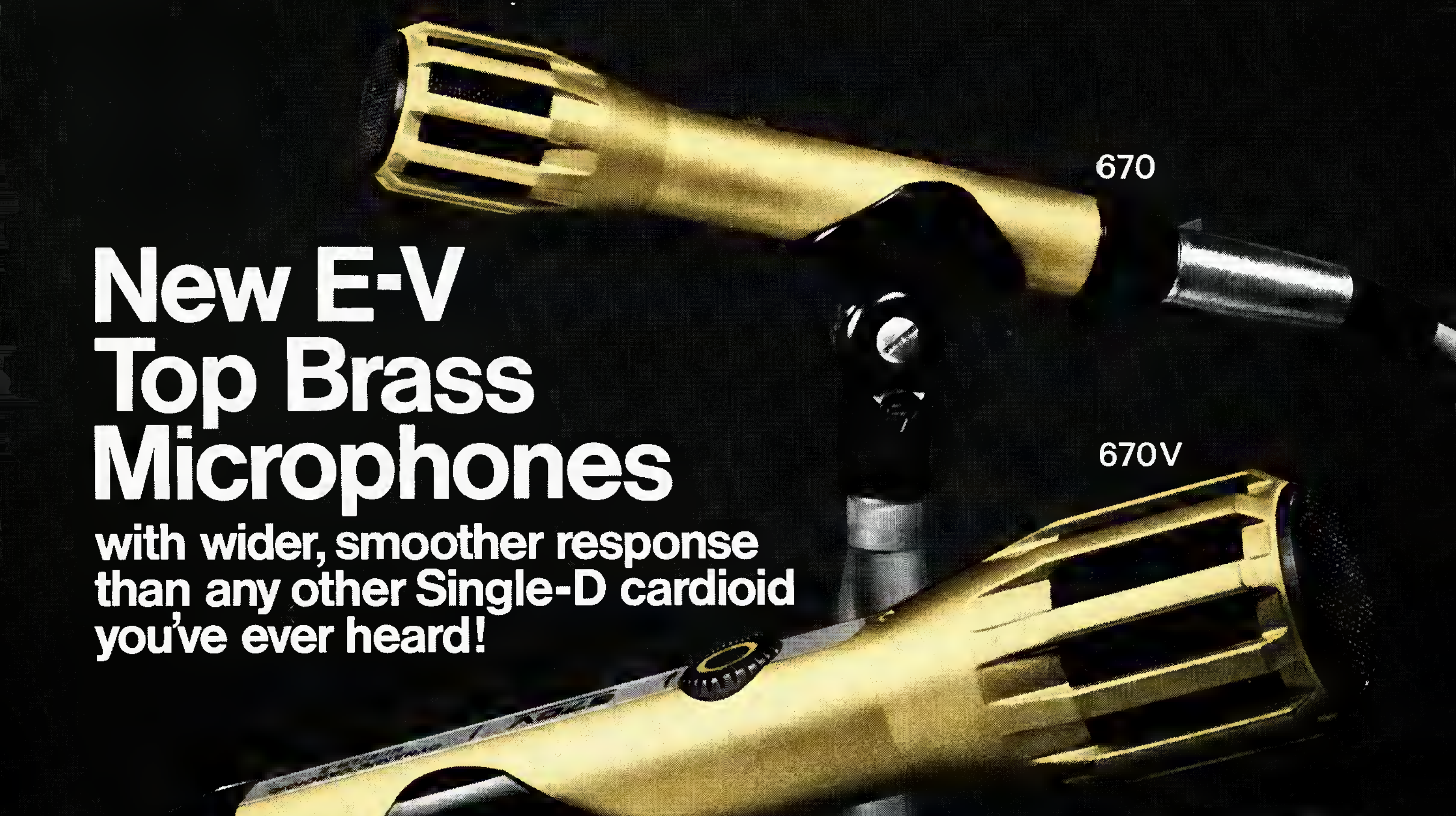

Exciting development of pioneering laser holographic research.

Inside the E-V microphone laboratories is a small, dark, quiet room. And in the room is a sophisticated holographic camera. Using newly developed techniques, E-V engineers can for the first time study microphone diaphragms under actual operating conditions. Movement as small as .000005″ can be seen and precisely evaluated. And this unique facility has led to new insights into microphone behavior and specifically, a brilliant new microphone design. The Model 670.

Inside the 670

Let's take a look inside the 670 to see what makes it uniquely better than any other Single-D dynamic cardioid (including our own earlier designs). The big difference is the diaphragm itself... which functions closer to the theoretical ideal than any other yet measured. It starts with the familiar Acoustalloy® sheet, formed to provide compliance rings at the edge, but absolutely flat in the center. Bonded to the center is an odd dome, made of compressed polystyrene, that looks much like a squashed aspirin tablet.

This unusual construction adds high stiffness with optimum mass. High frequency diaphragm breakup is eliminated. At 24″ working distance the 670 has virtually ruler-flat response from 60 to 14,000 Hz.* This makes possible a host of benefits: a pure cardioid pattern with virtually no lobes, less chance of feedback or overloading of inputs at high levels (since peaks are gone), and — of course — superb sound plus excellent rejection of unwanted noise.

Rugged Internal Construction

The "head" of the Model 670 is built as a unit (and is actually field replaceable). The head is suspended within the case using a pneumatic shock absorber that soaks up impact noise while protecting the assembly. All resonators are built in for added stability and reduced sensitivity to shock. And the entire head is assembled without glue, so that structural integrity is maintained despite extremes of heat, humidity or mechanical shock. The case itself is machined from a solid bar of aluminum, then anodized in the Top Brass finish for lasting beauty.

In every detail the 670 is outstanding. For instance, you can easily change from Hi-Z to Lo-Z impedance without soldering. And the professional connector insures minimum maintenance. Our pop and blast filter is unusually effective, a result of computer-aided research. Even the volume control on the Model 670V is unique. Designed for easy one-hand operation, yet placed to minimize accidental movement.

Light Weight, Low Cost

With all these features, you might expect a heavy microphone... and a big price tag. Good news: the Model 670 is half the weight of competitive Single-D microphones and is priced to please your pocketbook. Just $76.50 list for the Model 670 with on-off switch and $83.50 for the 670V with built-in volume control. Cable and stand mount is included. With standard phone plug add $3.50 list. Carrying case $3.50 list.

Find out for yourself the real benefits of E-V research. Choose the new E-V Model 670 or 670V Top Brass microphones. They're at your E-V sound equipment headquarters now.

*Of course as you move close, bass response is accentuated, a basic characteristic of all Single-D cardioid microphones.

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COMING NEXT MONTH

• In June, our camera visits Colonial Williamsburg in Williamsburg, Virginia. In the eighteenth-century setting that is Colonial Williamsburg, a behind-the-scenes activity goes on. Included are audio-visual displays, a sound motion-picture capability, and a recording studio. A number of commercial releases of eighteenth century music, recorded on location and using original instruments have been made here.

The Audio Engineering Society's Los Angeles Convention is covered by a *Picture Gallery* that shows the latest consoles, recorders, and equipment that was displayed for the visiting audio professional.

The *Technique of Electronic Music* is the title that Robert Ehle has given his work. It will appear in two parts for June and July and will provide the working audio engineer with much of the basic information he needs to understand this fast-growing field.

The final installment of Melvin Sprinkle's *Acoustics for Audio Men* will appear.

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

ABOUT THE COVER

• Our unusual cover this month is the inventive creation of Eastern Sound Co., Ltd., of Toronto, Canada. This new facility incorporates, among other items, the Ampex MM1000 and a custom console by Automated Processes. The photo above shows, in a more conventional way, the control room.

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letters

The Editor:

There is to a high degree a tendency in advertisements to emphasize the importance of single parameters well-knowing that as regards engineering the best result can be achieved only by an expedient adjustment of each parameter in proportion to each other. A consumer organization ought to treat the whole complex in a more objective way.

The Consumers Union report on stereo phono cartridges judges quality on *one* criterion only, that is wear on records when tracked with the minimum vertical force that a particular cartridge is capable. It is relevant to point out that CU does not comply with its own judgement of what represents quality, nor does it take the necessary steps to ensure that its methods of testing do indicate which of these cartridges fulfil their criterion of quality.

It can be easily shown that for two otherwise identical cartridges, where one has a spherical (or conical as it is sometimes called) and the other an elliptical stylus tip, the elliptical will show more wear than the spherical although the minimum vertical tracking force (MVTF) is identical. By CU's own standards all elliptical cartridges should be rated below their spherical counterparts, but in fact CU "judged elliptical styli, as a group, slightly preferable to spherical styli." CU does not mention higher wear with elliptical styli, but purports to take wear as the basis of quality in its tests.

"To assess comparative distortion from record wear that would result from differences in vertical tracking force, we put identical discs through a series of playings with two pick-ups—one pick-up requiring a VTF of 0.7 (meaning MVTF), the other a VTF of 2.5 grams", comments CU. Such a test is irrelevant in testing the effect of vertical tracking force on record wear.

MVTF is dependent on many factors. Among them effective stylus mass, compliance, tone arm design, bearing friction on one side and groove modulation, diameter and record surface etc. on the other. Many of these factors also determine record wear although wear due to each factor gives rise to different kinds of audible distortion.

A modern high-quality gramophone pick-up is a compromise between many such factors, among which are VTF and record wear, which contrary to CU's conclusion, are not synonymous. The test on which CU's conclusions are based only proves that the particular cartridge used, with a VTF of 2,5 grams, shows more wear than the particular cartridge used, with a VTF of 0,7 grams. To test the effect of tracking force on record wear all other factors would have to be held constant, while only the tracking force varied. Thus identical cartridges in identical tone arms playing the same record with different tracking forces would show the effect on wear of tracking force alone if finally listening tests were made with the same cartridge. If such a test is carried out, it is seen that for cartridges with VTF under 2 grams there is negligible additional wear as the tracking force is altered from below 1 gram to over 2 grams, and that wear will occur in exactly the same parts of the record for the two tracking forces. Of course, if the pick-up is tracking with a force below the MVTF of the cartridge, there will be more wear at this force than at the higher tracking force. It can also be shown that wear characteristics for two cartridges with the same MVTF playing at the same tracking force can be widely different, or that some cartridges with a lower MVTF have more audible wear in the inner grooves than some with a higher MVTF when playing at the same tracking force. This is because in these cartridges MVTF is determined by different factors.

The CU tests indicate that wear was due to a higher tip mass in the cartridge used as wear was most

advertisers index

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THE SOUND ENGINEERING MAGAZINE

Good 16-track masters don't just happen.

Unless the Dolby System is used, the final stereo master may be only marginally quieter than an 8-track stereo cartridge.

New sixteen-track recorders are essentially limited in performance by tape noise. No matter what tape is used, noise increases by about 9 dB when the sixteen-track original is mixed down to a two-channel stereo master. This is equivalent to cutting track width down to that of an 8-track cartridge and reducing tape speed to $7\frac{1}{2}$ ips. What a waste of time, money, and effort.

The Dolby System eliminates all this. The noise level of a Dolby recording, even when it is reduced to stereo from sixteen tracks, can easily be better than that of a two-track original recording on the same kind of tape without the Dolby System. At the same time, print-through and cross-talk are also reduced by 10 dB, keeping stereo placement exact and silent passages velvet quiet. Sessions move faster because setting up pre-equalization is unnecessary. Instead, equalization can be worked out during mixdown without affecting the sixteen-track original and without watching the second hand of the clock as session time ticks away. Nor is there need to ride gain on dead channels during mixdown to keep out noise; the Dolby System takes care of that.

The Dolby System makes recording and reduction easier and faster, with time for more attention to creative values, less to technical problems. It makes good engineering 10 dB better.

A mixdown at John Mosely's Command Studios, London. Command, like every other London 16-track studio, is Dolby equipped on every track.

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marked in the inner grooves.

High frequencies become more difficult to trace on this part of the record, whereas low and medium frequencies are, relatively, equally difficult to trace over the entire record surface, and wear would be expected to be even for equally loud passages everywhere. It is possible that the high MVTF of this cartridge was due to the high tip mass, but this is identical. It is equally possible that its low and medium frequency tracking was up to the standard of the cartridge with the low MVTF, but this is also irrelevant. It is important to understand that the difference in wear in CU's test was due to causes other than the vertical tracking force used.

A pick-up tracking high frequencies subjects the groove walls to enormous forces, in fact, several hundred times the vertical tracking force. In order to reduce distortion due to mistracking the effective mass of the moving parts has to be made smaller.

A pick-up with a high effective tip mass (ETM) distorts high frequencies more than one with a low ETM, even on the first playing of a record, especially on the inner grooves. Increasing vertical tracking pressure may reduce audible distortion as mistracking occurs over a small interval, but wear remains largely unaltered because forces to which the groove is subjected hardly alter.

In the design of a good cartridge both MVTF and record wear are factors for the designer's consideration, but others, such as frequency response, cross talk, inherent distortion, transient response, pulse response, phase shift etc., are equally important for the sound quality. To grade cartridges on a scale with MVTF as the only criterion of quality, when this factor does not even determine record wear, is as irrelevant as grading them only on frequency response, tip mass, (motional impedance?) or any other single design specifications."

E. Rorbaek Madsen
Chief Engineer
Bany & Olufsen
Struer, Denmark

The Editor:
I read with interest John Woram's column *The Sync Track* in the January issue of **db**. His points about console automation are certainly well taken. We had essentially the same thoughts when we began development of the Olive Series 2000 console components.

We have used the voltage control principal mentioned by Mr. Woram for several functions. Such variables as fader positions (inputs, subs, masters, and echo returns), compression, limiting, and keyable gain reduction are all externally programmable by voltage control. This technique permits these functions to be automated on a dynamic basis using our Automated Remix Programmer. The technique also lends itself to the gradual gain reduction spoken about. The master faders or the sub master fader require only a single ramp generator synced to an elapsed time indicator.

As far as the noise gate is concerned, we have such a device in our input module, as mentioned in the column. A number of additional features permit substantial flexibility in this device. A multiposition switch on each input allows bussing the gates together with up to 10 busses possible. A switch selects mode of operation, *i.e.* gate to close with signal or open with signal. It is possible to have one or more inputs control themselves and/or one or more other inputs. For example, a lead trumpet could open other trumpet mics and close or reduce the gain on drum mics.

The compressor/limiter on each input can also be linked in this manner.

As far as punch-ins are concerned, our monitor mix and sync module contains all the logic and routing circuitry to "automate" the switching required when changing from playback to sync to record modes. In essence it interfaces the console with the multitrack tape machine.

It was our objective in automating these features to free the engineer/producer from some of the time consuming mechanical tasks of recording or mixing and provide him with a new creative tool by being able to control more dynamic events at one time. The more the creative engineer can be freed from the constraints imposed by technical requirements, the better can be the ultimate product. As newer recording techniques are introduced, more tracks made available, and greater use is made of quadraphonic sound, the need for automation becomes apparent—not to supercede the recording engineer, but to perform for him the switching, patching and other distracting set-up tasks that are a part of multitrack recording.

The less occuped the engineer's mind is with the mechanics of the session, the more he is able to listen to what he is producing—which is, after all, what really matters.

Wayne Jones
President
Olive Electro-Dynamics Inc.
Montreal, Canada

NORMAN H. CROWHURST

THEORY AND PRACTICE

• Not long after last November's issue appeared, about input matching, I received a letter from a reader gently suggesting that I had not told all, and possibly not the most important part, on that subject. In that issue, I opened by kind of dismissing input transformers as an input device, and getting almost immediately into circuits that enable transistors to provide appropriate matching.

That struck me as a little ironic, because I could dig out quite a few articles that I have written, saying almost the same things this reader was telling me: how input transformers do not match in the classic theoretical sense, being designed for a high primary impedance, to avoid the 6-dB matched-impedance voltage loss. At least that was the reason he gave.

This is also a little ironic, because some time ago, I found myself in a sort of argument about the then proposed IEEE standard definitions about insertion gain, in the context of which this talk about 6 dB matched-impedance voltage loss would have been heresy! It seems that in audio, as in religion, whatever you say, someone is ready to disagree with you.

As each has some valid points, *taken in the correct context*, this is a real theory and practice situation. So maybe I should recap some of those earlier bits of theory and practice, that must have been written before this column got started, and proceed therefrom, with the hope that I can satisfy everybody!

First, that 6 dB "voltage loss". It is simple enough really, but our semanticists are apt to argue about whether we use the right words to describe what happens. In classic matching theory, a source of signal consists of a voltage source with an associated impedance. To make the numbers round, if non-standard, assume that the voltage source is 1 volt, and the impedance is 250 ohms.

When this is matched to another 250 ohms, into which the signal is delivered, this gets 0.5 volt (*Figure 1*). And 0.5 volt delivered into 250 ohms represents a power of 0.25/250 watts, or 1 milliwatt. Thus the transfer level, under an idealized matching arrangement, where the two impedances match, is 1 milliwatt, or 0 dBm.

Now, the 6-dB voltage loss to which my friend referred is the fact that the 1 volt within the source, whatever that is, gets reduced to 0.5 dB actually delivered to the 250-ohm input. But whether that is a loss or not depends on some other things about the circuit. If, *without anything else changing*, voltage level changes in a ratio of 2:1, that is a level change, gain or loss, or 6 dB.

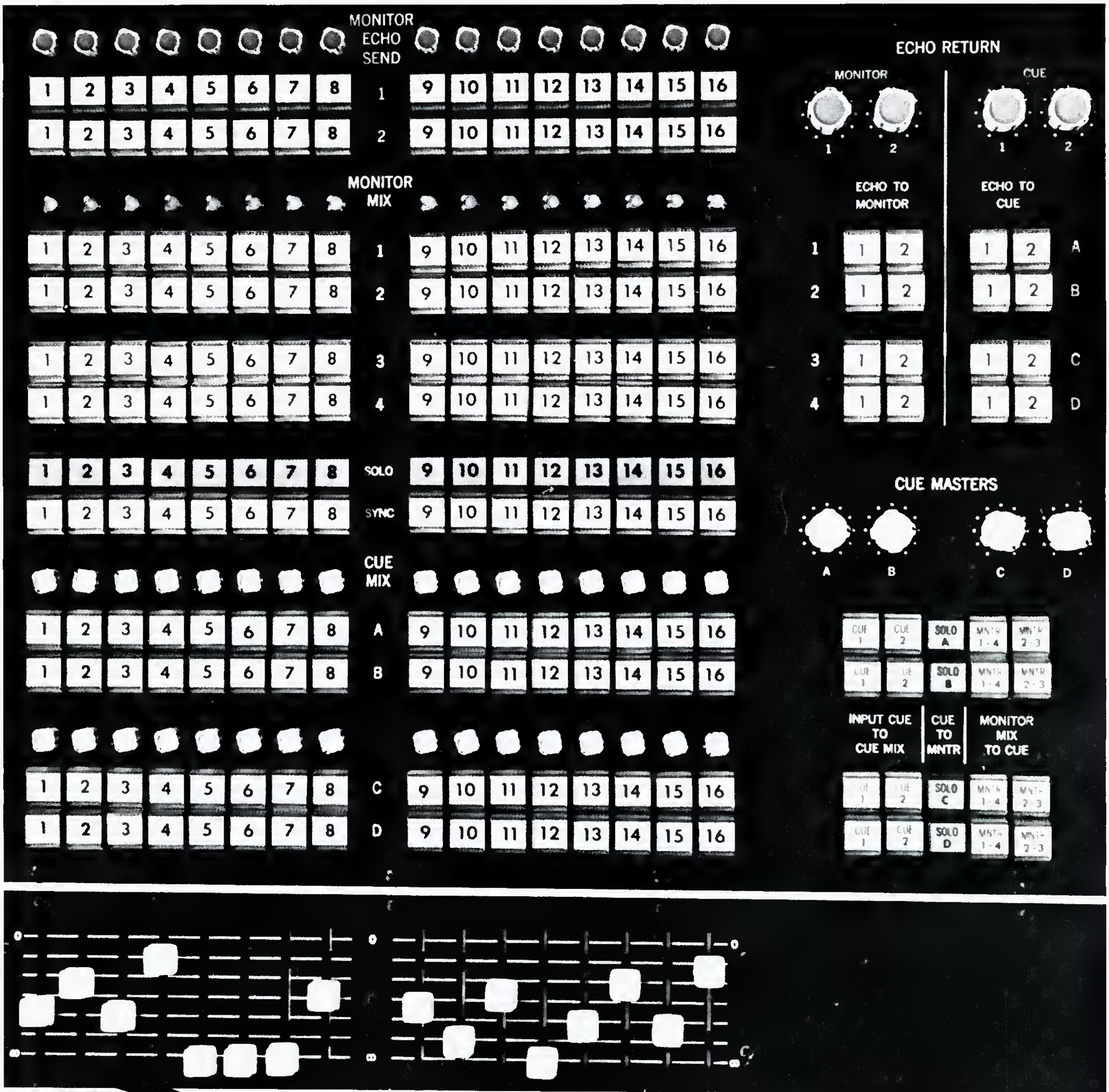
Let us go back for the moment to tube days, which is somewhat similar to the use of *fet*'s, for those who do not remember those funny glass bottles with hot cathodes and things that produced a different kind of amplification from transistors. The grid, which was the input electrode, of a tube presents a very high impedance—in the megohms—and possibly some input capacitance—some 50 to 100 micro-microfarads (all right, the good ones could be a lot lower than that).

Now, if you connected a 250-ohm source to the grid input of a tube, its voltage would appear at the grid, undiminished, and connecting a 250-ohm resistor across that, just for matching, would really lose 6 dB. No question!

But who in his right mind would do that? Amplification really concerns power, and the grid actually presents an impedance of, say 1 megohm (probably a lot higher than that, but the figure will do, for illustration). Putting the full 1 volt across 1 megohm represents a power level input of 1 microwatt, or −30 dBm. Putting another 250-ohm resistor across the 1 megohm will not deliver any more signal to the grid—6 dB less in fact. The input level *to the grid*, which is what gets amplified (the 1 milliwatt in the 250-ohm resistor is lost forever, as a minute amount of heat), has dropped from −30 dBm to −36 dBm.

What people of those days usually did was to employ a transformer. Typically, this would present a step-up of about 20:1, from 250 ohms to grid. Now, a voltage ratio of 20:1 represents an impedance ratio of 400:1 (20 squared) because voltage steps up and current steps down, by that ratio.

So to produce a power match now, the secondary of the transformer (*Figure 2*) should be loaded with 400



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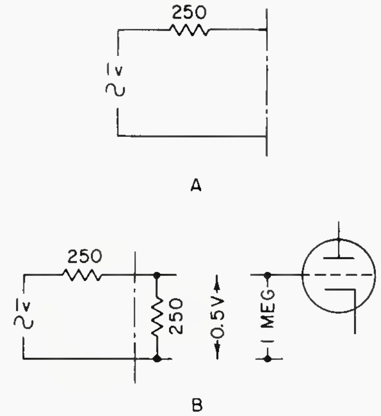


Figure 1. The classic matching concept requires a 250-ohm source (A) to be matched into a 250-ohm load (B) which reduces the available voltage from 1 V to 0.5 V.

X 250 = 100,000 ohms. Then the primary will receive 0.5 volt, and the secondary will get 10 volts. Across 100 k, this also is 0 dBm, but across 1 megohm it is −10 dBm. That is a little better, and provides an opportunity for keeping the signal further above noise (not that a 0 dBm signal is in any trouble, but at lower relative level, the improvement of 26 dB is worth while).

But now, leave that 100 k off the secondary, so it is loaded only with the 1 meg grid impedance, and the voltage rises (approximately) to 20 volts, because the primary will get 1 volt, instead of 0.5 volt. Now the input level is about −4 dBm.

As well as getting that 6 dB extra, raising the resistive load on the transformer will improve its high frequency response, almost for certain. Now, if that 250-ohm interconnection links a 250-ohm source (say the output from a line amplifier to the input of a power amplifier) on the same equipment rack, in the same studio, or at least in the same building, we have said about all there is to say about this little situation. The interconnecting line has a 250-ohm impedance at the input end of it, so anywhere along that line its impedance will measure 250 ohms—even at the output end, where it connects to this input transformer.

But if the interconnecting link is much longer (say a telephone line from one city to another many miles away) the situation changes. Such a line has a characteristic impedance of its own, determined by its inductance and capacitance per unit length (*e.g.* per mile). Failure to terminate such a line by a matching impedance at its receiving end can result in reflections: some of the signal will get reflected back along the line.

Terminating the 20:1 transformer with 1 megohm will reflect that value divided by 400 on the primary, which is 2.5 k, instead of 250 ohms. Suppose the transmission time along the line is 200 microseconds. This means that the line is 1 wavelength long at a frequency of 5 kHz. Terminated by an impedance 10 times its characteristic impedance, this line will reflect that at

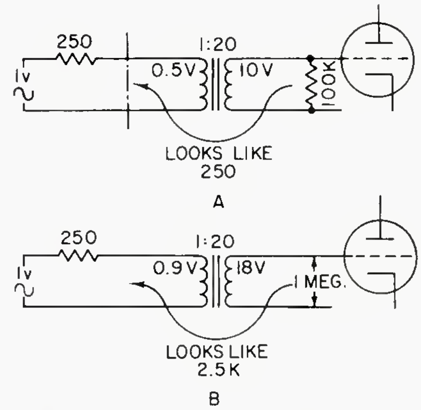


Figure 2. Use of an input transformer in a tube amplifier: (A) using the idealized classic matching, through the transformer; (B) using only the input impedance of the grid circuit. The latter results in a voltage gain of somewhat less than 6 dB.

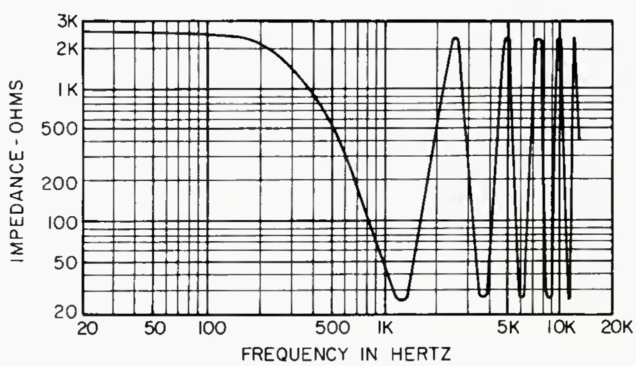
even half-wavelengths from the termination. But at odd quarter-wavelengths, it will reflect an inverse: 250 ohms divided by 10, or 25 ohms.

So at the sending end, this length of line will have a fluctuating impedance, which is 25 ohms at 1250, 3750, 6250, 8750, 11,250, 13,750 Hz, etc., going to 2500 ohms at 2500, 5000, 7500, 10,000, 12,500 Hz, etc. Can you imagine what kind of a crazy frequency response that will generate, fed from a 250-ohm source? It's shown at *Figure 3*.

Of course the length and its associated transmission time will vary the spacing of the bumps and dips. If it is short enough, they will all be well enough above the audio range, which is why this arrangement is *ok* for around the building or studio.

But if such an input transformer is used under this condition, the line must be correctly terminated. To avoid the high-frequency loss associated with secondary-loading the input

Figure 3. The reason for classic matching—and when it is important; when the input terminates as considerable length of line. Here is the impedance reflected to the sending end, when a line with characteristic impedance of 250 ohms, and a transmission time of 200 microseconds, is terminated with 2.5 k, the value presented by circuit in Fig. 2 (B).



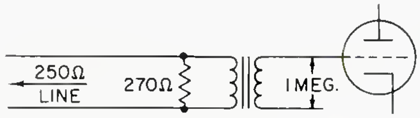


Figure 4. A better way to terminate the line, where this is necessary, is to do most of the matching on the primary of the transformer, as shown here.

transformer, the primary side can be loaded so the line is correctly terminated. As the 1 megohm secondary load looks like 2.5 k on the primary, a resistor of about 270 ohms will produce a 250-ohm termination for the line (*Figure 4*).

There is not enough space to extend this consideration into transistor applications in this issue, so that must wait for another time. I can understand the reader who wrote to me, well enough. It's a long time since I explained anything like the above, so his comment came at a good time to remind me.

What puzzles me is the people who pick over definitions like voltage loss and gain. Are they being fussy, or don't they know what we mean? I am well aware that the whole thing can be treated by elaborate theory (or it seems so to me) of insertion gain or loss, working with ideally matched impedances throughout, and then correcting with impedance mismatch losses where impedances do not match.

The problem with that is that it assumes we are always working with power gain and loss. Unless we are very careful with our terms of reference, that first instance in *Figure 2* would calculate as causing a mismatch loss of 20 dB, instead of the actual gain of 6 dB, as compared with connecting the 250-ohm termination. Tricky, to say the least. ■

back issues available

A limited number of back issues of db are available to interested readers who may have missed or misplaced earlier issues. When ordering please indicate date of issue desired and enclose 75c for each copy.

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John M. Woram

THE SYNC TRACK

• In reviewing last month's issue of **db** I noticed that in the wealth of information pertaining to the microphone, two things were given little or no attention; the ribbon microphone and the *stereo* microphone.

THE RIBBON MICROPHONE

Everyone seems to have an opinion about the relative merits of dynamic *versus* condenser microphones. But, somewhere along the way, the ribbon microphone has fallen into partial obscurity. This is understandable since—no matter where you stand in the dynamic/condenser argument—you're probably ready to concede that the condenser has a higher output, and the dynamic will take more abuse, than the typical ribbon mic. These characteristics have made them eminently suitable for the various requirements of the modern recording session, and the ribbon has lapsed into a very poor third place.

Except by Johnny Carson. Although the orchestra uses condensers, he has a Shure SM-33, a ribbon mic, on his desk. Most ribbon mics have figure-8 patterns, but this one happens to be a super cardioid, which I suppose makes it ideal for the needs of the *Tonight Show.* I haven't got the foggiest notion how one gets a super-cardioid pattern from a ribbon. In thinking it over, I've just about concluded that it cannot be done, which means one of two things. Either I'm wrong, or there is no such thing as an SM-33.

The smart money is betting of the first alternative. In fact, I now have my own SM-33, which I suppose somewhat weakens my argument against its existence.

A little while ago, I set it up next to a snare drum and compared it with my favorite condenser microphone. It really sounded terrible, which didn't surprise even me, since the condenser is so well suited to respond to the sharp attacks of the snare.

But on vibes, or marimba, the mic really proved its worth. The vibraphone in particular is an elusive instrument to record, especially during a noisy rock-type session. It's a wide sound source, rich in harmonics, and not particularly loud. And more often than not, the musician is doubling on other percussion instruments, and wants to be as close to the drummer as possible. But the closer you mic the vibes, the more you favor just those few notes directly in front of the mic.

With the SM-33, you can back off enough to cover the entire instrument, yet keep off-axis leakage to a minimum. At 90 degrees, the response is down about 5-8 dB more than many other cardioid types. And that *warm, mellow* sound characteristic of most ribbon mics seems just right for a vibe pickup. It's also great for toning down an overly shrill electric organ, and because of its physical shape, makes a handy microphone for the harmonica.

With the exception of this particular microphone, and another one which I'll get to in a minute, I haven't used any modern ribbon mics, and in fact, don't recall hearing of any. The old RCA 44bx and 77 were ribbons, but I think both have long since been discontinued.

The point of all this is that although most manufacturers seem to be concentrating on either dynamics or condensers, the ribbon microphone should not be forgotten entirely, and it would be well worth it to have at least a few around.

STEREO MICROPHONES

The Bank & Olufsen BM5 is an interesting ribbon microphone. Actually, it is two figure-8 microphones in one housing. The upper ribbon may be rotated through 90 degrees with respect to the lower ribbon, and the outputs are electrically independent of each other. The entire microphone costs $99.95, which makes it a very attractive way to find out more about stereo microphones.

For some reason, the stereo microphone is not as popular in America as it is in Europe. Often stereo microphones are thought of in terms of matrixing transformers and sum-and-difference signals. However, there is no reason why a stereo microphone cannot be used as two (electrically) separate microphones that happen to share a common housing.

When stereo miking any large sound source, such as a string section or a chorus, the engineer will probably use two (or more) mics, spaced some distance apart. Often there are cancellations due to the interaction between the mics, or if they are positioned in close, you may get a "multiple mono" sound, with no really defined center information.

A stereo microphone may often be used to advantage in these situations, either by itself, or as an over-all pickup, with a few accent mics in close. The stereo effect is excellent. Some months ago, I had occasion to record a rehearsal of a rhythm and brass group in a church. Augmenting the group was a large chorus and soloists. Since the recording was done on short notice, and was intended only for auditioning the music, there was neither time nor need to assemble a sophisticated recording set-up.

A good thing too. With a little more time, I probably would have botched up the whole thing. However, I just hurriedly set up the BM5 in the middle of the church, pointed one ribbon to the left side of the altar, and the other to the right, and sat back trying to look as though I was being very clever.

It turns out that I was. On listening to the playbacks later on, we were all impressed with the really fine sound, and there was no lack of directionality. I'm sure I couldn't have done better with a multi-mic pickup. Of course, I was working in an excellent acoustical environment which greatly enhanced the recording. However, even in a dead studio, a stereo microphone may sometimes be used to advantage. When double miking a piano onto two tracks, a stereo microphone may eliminate that hole in the middle sound. And if you want to place a group in a tight semi-circle, its a lot easier to get a good balance between voices, since the singers have apparently only one microphone at which to work.

There's no reason why you can't fashion your own stereo microphone by taking two identical microphones and rigging them up so that their elements are close together and about 90 degrees apart. However, it's rather a nuisance, and if you do stereo recording (different from multi-track mono) you may find the stereo microphone a sound investment. ■

classified

Classified advertising is an excellent and low-cost way to place your products and services before the audio professional. If you are a prospective employer seeking skilled help or an employee seeking a change you will find that the classified pages of **db** reach the people you want. Special low rates apply for this service.

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3 times — 10%

6 times — 20%

12 times — 33%

Agency discounts will not be allowed in any case.

Closing date for any issue is the **fifteenth** of the **second** month preceding the date of issue.

Martin Dickstein

SOUND WITH IMAGES

• Until recently, there was quite a difference in the color and contrast observed when a film was seen projected onto a front screen and when seen either on the air or on a t.v. film chain system. This proved to be very disquieting to both the advertising agencies and their clients who wished to see the film properly before sending out copies to t.v. stations but did not have the film chain facilities for screening.

Recognizing this problem, a good deal of research was done by the Eastman Kodak Company and the Canadian Telecasting Practices Committee, and each came up with recommendations which varied slightly from each other but agreed to a great extent on the requirements of a front projection system which would simulate t.v. viewing without the use of a color film chain. In accordance with these findings, the SMPTE has also published a recommended practice for use in the setup of a t.v. viewing system.

Normally, in front projection, a 16-mm film is shown with an incandescent light source whose color temperature is in the 3200 deg. K range. This projects a warm reddish appearance in the image and the film is usually balanced during the developing stage toward the blue to compensate. By contrast, a 35-mm film is shown with a xenon or arc source and shows up more in the blue range. Thus, if a 16-mm film were shown under the same circumstance, it would appear washed out but also much too blue. However, the 35-mm film, normally projected at a 5400 deg. K color temperature is already better balanced for t.v. transmission. Both reports agreed that the color temperature of the 16-mm film projection system should be closer to that of the 35-mm and the suggested color temperature is 5400 K ± 400 K to simulate the color temperature of the t.v. monitor.

Another difference in seeing a film in a front projection system, rather than in a film chain distribution to a t.v. monitor, is that a screening in a projection room requires that the lights be turned off while t.v. is usually viewed with a significant amount of ambient light present. In the darkness, the eye will adapt to the ambient condition and will accept color variations that may appear completely intolerable under the lighting conditions found in t.v. viewing where no eye adaptation is necessary. Therefore, to simulate proper t.v. viewing conditions in a t.v. film previewing room, it has been recommended that the ambient light in the room be adjustable to provide a color temperature in the range of 5000 to 6500 K (the same as the viewing screen) and the surround light at the screen location be no more than 3 ftL.

It will also be remembered that the recommended screen brightness for a front projection of a 16-mm film is 16 ftL. (± 2). It was found that the t.v. screen, however, had a brightness of about 40 ftL. (Both measurements are taken with a film projector turned on but without any film in the aperture.) To compensate for this, the recommended screen luminance during projection of a t.v. film is 40 ftL. (± 4). Thus, with normal density film going through the projector, the screen brightness during front projection will be cut down to about half and the t.v. screen luminance will correspondingly be diminished to about 20 ftL. Both reports agree that at a distance of 5 per cent of the screen width from the side edges of the screen, the luminance should be 80 ± 10 per cent of the central luminance, both values measured on the horizontal axis. (This is in agreement also with the values recommended for a normal 16-mm front-projection screen system.)

As a further enhancement of the illusion of watching a t.v. set during this special front projection setup, the screen should be masked down to the size of the t.v. screen being simulated. Thus, for a 25-inch set, the mask should be 16 high x 20 7/16-inches wide. For a 19-inch set, the mask dimensions should be 12 x 15 5/16-inches. If a larger set is required for a larger audience, the mask might be 31 1/3 x 40-inches (note that W = 1.276H) but the Canadian recommendation is that a set larger than the 25-inches should not be used. The CTPC suggests that the audience be seated no closer to the screen than 4 times the screen height and no farther away than 6 times. Kodak gives the seating range as 5 to 8 times the image height. (Projected image dimensions for a 19-inch set are 12 3/8 x 16½-inches, for a 25-inch screen 16½ x 22-inches, and for the largest masking size 32¼ x 43-inches.)

The masking frame's outside dimensions are also recommended to provide a standardized viewing area. For a 19-inch monitor, the frame should have an outside width of 18½-inches and an outside height of 14 3/8-inches. For a 25-inch screen, the outside dimensions of the mask should be 24 x 18½-inches, and for a larger size screen correspondingly increased in both directions. The mask can be made of a large piece of ¼-inch plywood with the screen area cut out in the center to the proper dimensions. The area of the wood immediately around the screen, then, should be painted *flat black* to an outside dimension in accordance with the screen size and the corresponding mask dimensions given. The remainder of the wood should then be painted *flat light gray*. The reverse side of the framing mask should be painted completely flat black.

The decor of the room is also taken into consideration in the requirements for the design of a t.v. film viewing room. The wall behind the screen should be flat light gray; the ceiling, floor and side walls should be flat medium gray; and there should be only subdued colors in the furniture, etc., within the room to avoid any disturbance or distraction to the eye.

The screen itself is one subject of difference in the two reports and therefore the required output of the projector also differs. The CTPC recommendation is for a screen reflectance of approximately 20 per cent which closely conforms to the typical reflectance off a t.v. screen. For such a screen, the suggestion is made that a particular white (flat) paint be used as recommended by the National Research Council of Canada. (The name of the paint describes it as a "Velvet", and it is manufactured by 3M.) To provide the proper screen luminance from this screen, the projector should have an output of approximately 500 lumens for the 25-inch screen size and approximately 285 lumens for the

19-inch t.v. size. The recommendation is that lamp housings containing high-intensity light sources are required for such high beam outputs. (The low reflectance screen is recommended to prevent ambient room light from bouncing back toward the viewers and thus affecting the proper judgement of the projected image, and to resemble the appearance of the t.v. screen when not in use.)

Kodak, however, recommends the use of their newly developed Ektalite screen. This unit is made of a thin, specially grained aluminum foil rigidly laminated onto the surface of a spherically curved fiberglass shell and then mounted in a vinyl-clad metal frame. The result is a light weight screen with six to eight times the brightness of other front projection screens. The unit has reflection angles of 60 deg. in the horizontal and 30 deg. in the vertical planes. Beyond these points, the reflection drops to the level of a matte white screen. The screen is made 40 x 40 inches and can be masked off to the desired screen size.

The high efficiency of the screen allows it to be aimed to reflect the light from the projector as though the screen were a mirror. Thus, it can be mounted and tilted (it comes with a special mounting assembly for this purpose) to reflect the image directly at the audience seating area while the ambient ceiling light would be bounced down toward the floor in front of the viewers and thus prevent interference. With no image being projected onto the screen, and under normal lighting in the room, the screen will closely resemble the appearance of a turned-off t.v. set.

With the screen brightness so high during a presentation, it is suggested that it might be necessary to decrease the lumen output of the projector to achieve the proper screen luminance for t.v. simulation. To color-correct for the color temperature of the projector light source, a filter (Corning #5900), mounted in an adapter and afixed to the front of the lens, is suggested. Depending on the projector being used, the light source can be decreased by using a lower wattage lamp, removing the reflector in the lamp housing, adding on an aperture stop (with the filter) in the lens adapter, cutting down on lamp voltage (in projectors provided with this adjustment capability), or by using a slower lens. The final requirement of the system, however, is to achieve proper color temperature and screen brightness for the projected film image to appear as it would when transmitted on t.v. whether in a color film chain or on the air.

Not so incidentally, this same screen can also be used in a regular front-screen projection system with the regular projector light source and without the filter. The high reflective efficiency of the screen will permit keeping the room lights on at their normal illumination level rather than having to turn out the lights in a film or slide presentation. (This may upset the viewers who would normally fall asleep unnoticed.)

The results of both studies, if properly applied to a projection room system design and installation, will provide a very satisfactory t.v. simulation for those clients desiring a front-screen capability to permit previewing of t.v. films to be used for t.v. transmission (closed circuit or on the air) where there is no film chain facility available. ■

ARNOLD SCHWARTZ

THE FEEDBACK LOOP

• The discussion of time constants in circuit analysis continues from last month. We found that in an R-C or R-L circuit the time constant relates frequency response to the values of circuit components. We also posed a problem which is shown schematically in *Figure 1.* Here we have an amplifier with a 600-ohm output impedance driving a 1200-foot cable having a capacity of 60 μF per foot, and terminated with a 600-ohm load resistor. The question is, if it is required to have a response within ± 1 dB to 15,000 Hz, will the cable capacity cause any high frequency problems.

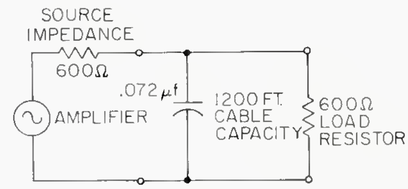


Figure 1. The effect of an amplifier driving a load through a 1200-foot cable.

Referring to *Figure 1* we find that the circuit resistance is equal to two 600-ohm resistors in parallel, or 300 ohms. The total cable capacity is $(1200) \times (60) = 72{,}000\ \mu F = 0.072\ \mu F$. The time constant of the circuit in microseconds is equal to $(R) \times (C)$, where R is in ohms and C is in microfarads. The time constant of this circuit then is $(300) \times (.072) = 21.6$ microseconds. Let's round this off to 20 microseconds. We know that 160 microseconds corresponds to 1,000 Hz; 20 microseconds is one eighth of 160 so that the frequency corresponding to 20 microseconds is eight times 1,000 Hz, or 8,000 Hz. Actually, since the time constant is slightly larger than 20 microseconds the corresponding frequency is proportionally lower than 8,000 Hz.

The response of the circuit is shown in *Figure 2* (solid line) where we have a 6 dB-per-octave slope which is 3 dB down at slightly below 8,000 Hz, and about 7 dB down at 15,000 Hz. Since the circuit response must not be more than 1 dB down at 15,000 our requirements are not met.

We know that with a 6 dB-per-octave slope the response is down 1 dB at one octave below the knee. If the response is to be not more than 1 db down at 15,000 Hz, then the 3-dB point has to be at least one octave higher at 30,000 Hz. The time constant at 30,000 Hz is easy to find. One of the two basic time constant-frequency relationships is 300 Hz and 520 microseconds; the time constant at 30,000 Hz will be 5.2 microseconds (5.2 is 1/100 of 520 and the corresponding frequency is therefore 100 times greater). If we had to decrease the time constant of this circuit from 21.6 microseconds to 5.2 microseconds we would have to either decrease the cable capacity or decrease the generator and load impedances. In practice, the latter alternative would be the only practical choice. If the input and output impedances were reduced to 150 ohms the total circuit impedance would be decreased by a factor of 4. This would give us a time constant of 5.4 microseconds and we would then have a circuit which was only slightly more than 1 dB down at 15,000 Hz.

If for some reason a decrease in cable capacity, or a decrease in circuit source and load impedance were not

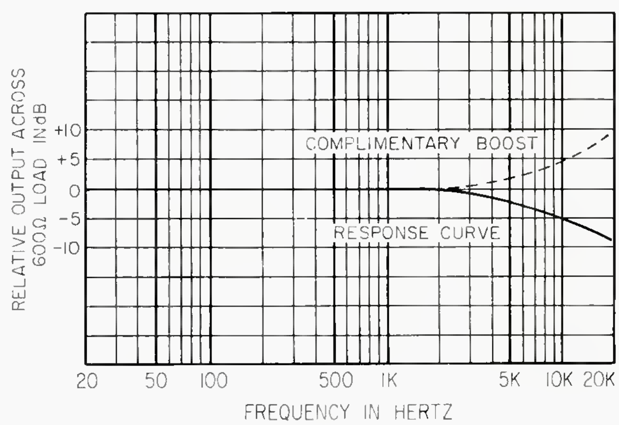


Figure 2. The response curve of the circuit shown in Figure 1.

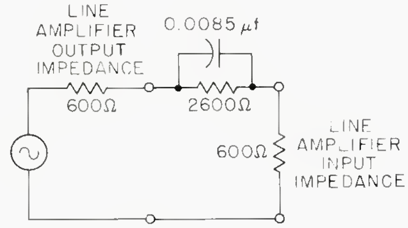


Figure 3. This complementary boost circuit will give you the correction shown in Figure 2 if it is inserted between line amplifiers.

practical solutions, we could compensate for the high-frequency attenuation by inserting a circuit with a response complementary to that of our given circuit somewhere in the transmission path. This complementary curve is shown as a dotted line in *Figure 2.* A simple R-C compensating network is shown in *Figure 3* which can be designed to compensate for the high frequency attenuation. For purposes of illustration of R-C circuit calculations we simply insert the network between two line amplifiers each with 600-phm input and output impedance.

The attenuation region of interest extends for about one octave, from slightly below 8,000 to 15,000 Hz. Since the slope is 6 dB-per-octave the total attenuation, and therefore the total boost required for the compensating circuit, is about 6 dB. However, since the beginning and end of a 6 dB-per-octave slope is quite gradual, we have to use about 10-dB total boost in our compensating network to insure that we have a reasonably accurate 6 dB-per-octave slope in the 8,000 to 15,000 Hz region. For this reason we select a series resistor that will cause a 10-dB attenuation at low frequencies. At high frequencies the resistor will be "shorted out" by the capacitor and provide our necessary boost. The series resistance (R) required for 10-dB attenuation is calculated as follows:

$$10 = 20 \log \frac{1200 + R}{1200}$$

$$10 = 20 \log 3.16$$

$$\frac{1200 + R}{1200} = 3.16$$

$$R = 2600 \text{ ohms}$$

The capacitor required to give us a boost of 3 dB at the required frequency is calculated as follows. The time constant of the circuit causing the attenuation is 21.6 microseconds. Therefore the value of (R) X (C), where R is the added 2,600 ohm series resistor and C is the capacitor shunting the 2,600 ohm resistor, should also be 21.6 microseconds. Performing this calculation, we get (2,600) X (C) = 21.6, and solving for C we get 0.0083 microfarads. The circuit will now provide a boost curve that is the complement to the attenuation curve up to 15,000 Hz. The curve will flatten out just above 20,000 Hz. If we rounded off the figures and used 20 microseconds the resulting circuit would still be within 1 dB of the required over-all response.

It may seem to some readers, not familiar with the time constant method of determining circuit behavior and circuit constants, that this method is somewhat roundabout. However, with a little practice and familiarization one can be as at home with time constants as with impedance. Many circuit problems need only approximate solutions. In these cases time constant calculations can be made very rapidly and with sufficient familiarity can often be done in your head—or should I say "while on your feet." ■

ROBERT HANSEN

Studio Acoustics

In this discussion of acoustics, the author discourses on the materials and practices that go into a studio construction—all from the practical viewpoint.

HOW do we get a studio, whether it is a recording, video, or broadcasting studio, built in the field? How do we do it without it costing us $100 a square foot; how do we do it in the space that we can live with, particularly if you are putting a studio into a high-rise building.

Let's make believe we are going to build a studio. Now do you prefer a broadcasting or recording studio—either way the first problem is finding the space to put it in. Our firm has been investigators, hunters, lawyers, psychologists, and everything else trying to find space for many types of studios. Broadcasting studios are generally put into high-rise office buildings. They present a lot of problems. If you are interested in having a recording studio, naturally you are looking for a place into which you can produce a lot of decibels. The main thing is that when you get a building you think you like—check it out, please. Don't be satisfied because it has 4, 5, or 6,000 square feet of space, a nice freight elevator, or the entrance is pretty. Check out who are your neighbors above and below. Check out who is next door to you. There may be a company with seven printing presses; if you visit on a Saturday when they don't work you don't realize the amount of vibration in the structure. Find out what kind of landlord you have and see if you can get something in your lease that may protect you from his noise making machines. He may have an old antiquated elevator system that should have been repaired twenty years ago that may start introducing very beautiful pure tones into your space.

The first consideration is: is the space really what you want from all aspects? Look out the window. If there is a water-cooling tower out there, you can build a wall or you can block the window up. That depends on what your lease says: if your lease says you can do anything but you will restore it when you leave, try not to do too much with this structure, because if you have got to rip it out it will be a problem.

Second, and this is another aspect, the cost; the square-foot price. Let's say you do find the space for a recording studio which is great—ten or twelve foot ceilings—you can do tricks with sound, deflecting or scattering.

If it is a radio studio, we try to live with nine feet if you can get it but duct work usually enters into it and the mechanical engineer always says "I need ducts that are 6 x 4 and you know he isn't going to get them under those beams. If he does, your ceiling will be 4½ feet. You have to be very strong. The only way to be strong is to hire yourself a very good architect, a guy that is very practical, knows the ins and outs and knows how to protect you from acoustical consultants like myself and mechanical engineers, both of whom are quite adamant people.

So you do need a good architect, don't try to do it yourself. I don't care how clever you think you are—you may design the most beautiful studio in terms of space and work flow—forget it. Get yourself an architect because he will tell you things you have never known; I have learned many things from architects. After you obtain a good architect, sit down and talk to him—he doesn't have to know anything about recording or broadcasting—what he will do is take the little bits of information you are furnishing him and translate those into construction material, put them on a drawing, and pump them into a specification.

Hope that he makes the drawing and the specifications nice and tight so that when you get a price from a contractor, you have the contractor where it hurts. Because

(continued on page 21)

Is the space really what you want?

. . . you do need a good architect, don't try to do it yourself.

Robert Hansen is a noted studio construction and acoustic consultant that has had among his clients the American Broadcasting Company, Westinghouse Broadcasting Company, WMCA, A & R Recording Studios, The Electric Lady, Sterling Sound, and the new Sears Building now going up in Chicago.

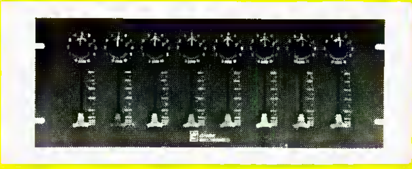
NEW PRODUCTS AND SERVICES

PROFESSIONAL SPEAKER LINE

• Exceptional sensitivity and efficiency is claimed for these speakers which have been produced for the music industry. Limitations inherent in conventional speakers have been designed out through an integral heat sink casting which radiates voice coil heat. Dual voice coil spiders, each 4.2 inches in diameter, offer optimum centering and prevent failure from cone fatigue. The finned Alnico V magnetic structure provides a flux density in excess of 13,000 gauss and an EIA-rated sensitivity of 54 dB. Power handling capability of the unit is rated at 200 watts, 1 hour, with a peak of 350 watts, 20 seconds continuous power input. Speakers are available in lead or bass versions with 12-, 15-, and 18-inch cones, and may be front or rear mounted.

Mfr: Gauss Electrophysics
Circle 73 on Reader Service Card

SUBMASTER MODULES



• This new addition to the Series 8 line of professional audio equipment is intended for submastering between the mixing busses and the multichannel recorder. It features i.c. operational amplifiers on each channel and has an overgain of 15 dB. Output is rated at +22 dBm into 600 ohms and noise at −80 dBm. The unit is available with Penny & Giles, Langevin or Kuhlman slide attenuators. The SM-8 and SM-8E differ in that the SM-8E has additional rotary controls for mixing echo signals into the program signals so that "WET" recording can be accomplished.

Mfr: Gately Electronics
Circle 82 on Reader Service Card

HIGH SPEED CASSETTE DUPLICATOR

• A new high speed cassette-to-cassette duplication system five times faster than any previous cassette duplicator has been placed on the market. The new model CD-200 master reproducer can duplicate up to 53 hour-long cassette programs in one hour using a single slave duplicator. The reproducer can drive up to five slave units for a total of 265 hour-long copies per hour. The system is designed for use by medium size contract duplication companies and by schools, libraries and businesses which require cassette copies of educational or instructional material.

Operating at a duplication speed of 75 in./sec., the CD-200 system makes copies 40 times faster than a cassette can be duplicated in simple real-time duplication. Other duplicators now in use operate at 15 in./sec., or just eight times faster than real-time duplication.

The higher speed is made possible by the use of vacuum servo columns in both the master reproducer and the

slave units. The tape is pulled out of the cassette into the vacuum chambers where the duplication process occurs. This results in close tape-to-head contact and assures gentle, precise tape handling despite the high tape speed. In addition, solid-state broadband electronics are used in the system to achieve the necessary bandwidth of 320kHz for high speed recording. The CD-200 system consists of a master reproducer and a slave unit equipped with an automatic cassette loader which holds up to 100 blank cassettes. As both sides of the cassette are duplicated in a single pass, a 60-minute recording is duplicated in 45 seconds at a tape speed of 75 in./sec. Rewinding the master cassette and the duplicate is done at 150 in./sec., which takes an additional 17 seconds. It takes another five seconds to eject the duplicated cassette and reload the slave unit with a blank cassette, for a total time of 67 seconds to duplicate an hour-long program and ready the equipment for the next duplication cycle. The system continues to cycle automatically as long as a blank cassette is in the loader.

Mfr: Ampex Corporation
Price: $9400 (two-track mono); four-track stereo—$10,500.
Circle 79 on Reader Service Card

ACOUSTIC DOOR

• Performance of this architecturally designed, flush, steel door is assured by cam-lift gravity hinges. These are sloped to lower the door as it is closed and to compress the acoustic seal firmly against the finished floor. When the door is opened, it is raised by the cam action of the hinges, and the compression seal is automatically released. Brought snug against the floor by the action of the cam-lift hinge as the door is closed, a continuous compression-silencer seal functions with the certainty of gravity. This action is described as a vast improvement over unreliable mechanical drop seals. Enhancing the acoustic effectiveness of the Noise-Lock compression seal are two sets of permanent magnets, one on the door leaf and the other on the frame. The magnetic action eliminates the need for noisy latch mechanisms. Furthermore, the door can be easily opened in a panic situation by pressing against any portion of the inside door-leaf surface. According to the company's engineers, gravity and magnetism together provide a better closure and a higher order of acoustical isolation, than either force could alone. The acoustical efficiency of this type of door is described as "high." Independent laboratory tests indicate an operating door assembly rating of STC 47; guaranteed field performance is STC 43. The door and its frame are primed to accept almost any covering—paint, fabric, vinyl, etc.—and the finish of the cam-lift gravity hinges is compatible satin chrome.

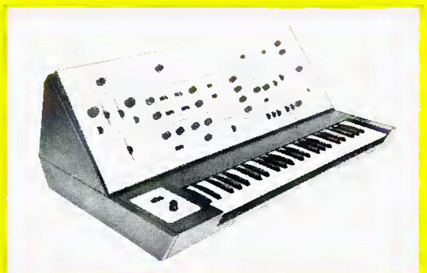
Mfr: Industrial Acoustics Co.
Circle 83 on Reader Service Card

FOUR TO TWO ENCODER

• The Electro-Voice professional encoder, a part of the company's "Stereo-4" 4 to 2 channel processing system is being marketed to studios and broadcast stations. The encoder, model 7445, is a standard 19-inch rack mounted unit and takes only 5 1/4-inches of space. Input and output lines are 600 ohms with zero insertion loss. The encoder causes virtually no degredation of signal response, distortion, or noise level. Using the encoder, recording studios can cut discs and tapes into a two-channel signal which contains both two-and four-channel information. Broadcasters can be on the air immediately with four-channel stereo information as existing stereo transmission equipment is used. No change in bandwidth or standards is required, and blanket approval has been received from the F.C.C. for the system to be used by any station. The company's home-type decoder is currently being offered for sale through high-fidelity outlets at a suggested list price of $59.95.

Mfr: Electro-Voice, Inc.
Price: $795.00
Circle 76 on Reader Service Card

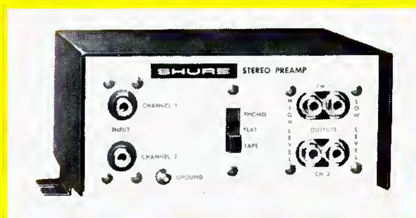
MUSIC SYNTHESIZER



• The Sonic V synthesizer is housed in an attractive contemporary cabinet and offers two-note performance capability through a four-octave keyboard, with a third input channel for microphone, electric guitar, tape, etc. All functions are controlled by rotating knobs or slide switches, which are mounted on a graphic flow panel to show the participation of each function in the resultant sound output. There are no patch cords. The manufacturer further states that the keyboard pitch level may be moved up, down, or compressed to produce microtonal scales, can be instantly reset to concert pitch (A=440), and is virtually unaffected by changes in temperature or humidity. The Sonic V measures approximately 34 in. wide x 21 in. deep x 13 in. high, and weighs about 40 pounds.

Mfr: Musonics Inc.
Price: $895.00
Circle 81 on Reader Service Card

AUDIO PREAMPS



• Two new stereo preamplifiers that provide the voltage gain, equalization, and choice of impedances necessary to operate magnetic phono cartridges and tape playback heads have been announced. The preamplifiers, the model M64 and the model M64-2E, provide complete freedom from microphonics, extremely low noise, and the ability to use 50 feet or more of output cable when used as an impedance converter and buffer amplifier. Both models have a single slide switch for selecting equalization for phono, tape, or flat. Positions provide RIAA for magnetic stereo cartridges, NAB for tape head equalization, and flat for microphone or for use as a buffer amplifier. Both models offer high level, high impedance outputs and low level, low impedance outputs plus a minimum of 50 db isolation between channels.

The model M64 operates on 120 volts, 50/60 Hz power line, or from an auxiliary 24 to 36-volt d.c. supply. The M64-2E is identical to the M64 except that it operates on a line voltage of 240 volts, 50/60 Hz.

Mfr: Shure Brothers
Price: $34.00
Circle 80 on Reader Service Card

CASSETTE TAPE TENSION MONITOR

• For the first time this M-100 cassette deck tension monitor permits direct measurement of the dynamic tape tension between the drive capstan and the two tape packs within a cassette deck. It was developed to aid equipment manufacturers and cassette users in obtaining maximum performance from both digital and audio decks. The monitor employs a special cassette which is instrumented internally with a strain gauge to measure tape tension directly. This cassette contains 150 feet of computor-grade tape, recorded with 1600 flux changes per inch to permit monitoring of speed changes. The cassette is connected with a meter which has ranges of 25 and 100 grams full scale. Power is provided by alkaline cells with a 400 hour service life. A direct output for oscilloscope observation is also provided.

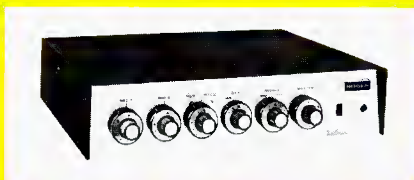
Mfr: Information Terminals Corp.
Circle 71 on Reader Service Card

TIME-SYNC GENERATOR

• A crystal time-sync generator for syncronizing a camera with the Nagra III or IV or Tandberg Model 11-1-P is now available. It is powered by the recorder and produces a 1.0 volt, 60 Hz synchronizing signal, with an accuracy of better than 10 parts per million which is less than + 1/6 frame in ten minutes. The syncronizing signal is recorded on the tape by the Neo-Pilot System, along with the audio signal.

Mfr: George Jensen (thru Ryder Magnetic Sales)
Price: $256.00 with cables
Circle 77 on Reader Service Card

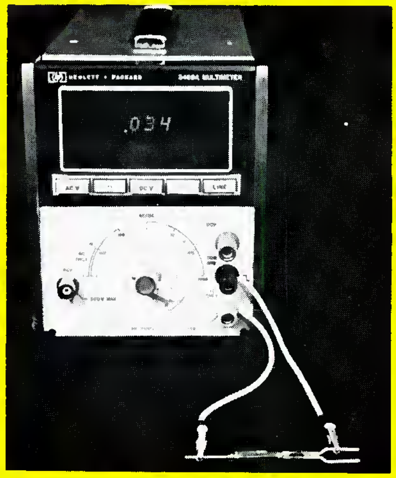
P.A. AMPLIFIERS



• Model C100 is representative of a new line of four public-address amplifiers offering 20, 35, 60, and 100 watts respectively. All are designed for continuous operation at full output from −20°C to +50°C. Protection against circuit damage from shorts or opens of the speaker lines. Two high-impedance mic inputs, each with volume control, two high-impedance high level aux inputs (again with faders) master volume control, bass and treble, and a power switch are provided. All mic inputs are filtered to guard against r.f. interference. Plug-in line input/line output matching transformers are available.

Mfr: Bogen Division
Price: (C100) $247.00
Circle 35 on Reader Service Card

WIDEBAND DIGITAL MULTIMETER



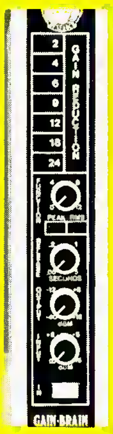
• A new multimeter with a digital display has high-performance features at a price near that of analog instruments. The model 3469A digital multimeter makes average a.c. measurements from 1 millivolt full scale to 500 volts over a frequency range from 20 Hz to 10 MHz, resistance measurements on its most sensitive range of one ohm full scale to 10 megohms full scale, d.c. measurements from 100 millivolts to 1000 volts full scale, and d.c. from 1 microampere to 100 milliamperes full scale. On a.c. volts, the model 3469A has an accuracy of from ± 0.25 per cent of reading ± 0.25 per cent of range to ± 2.5 per cent of reading ± 2.5 per cent of range, depending upon frequency and range. Response time is 2 seconds. The new multimeter measures 1 ohm full scale on its most sensitive range, to 10 megohms full scale. Accuracy of resistance measurements is ± 0.2 per cent of reading ± 0.2 per cent of range to ± 0.25 per cent of reading ± 1.0 per cent of range, depending upon the range used.

Accuracy of d.c. voltage measurements is ± 0.1 per cent of reading ± 0.2 per cent of range on the 100 millivolt range and ± 0.1 per cent of reading ± 0.1 per cent of range on the 1-1000 volt ranges. Input impedance is 100 megohms on the 100 millivolt and 1 volt ranges, and 10 megohms on the 10 volt to 1000 volt ranges. Overrange of 100 per cent of full scale is possible on all ranges except the 1000 volt range. Polarity is indicated automatically, and overload protection is provided on all ranges. Accuracy of current measurements is ± 0.2 per cent of reading ± 0.2 per cent of range.

Mfr: Hewlett-Packard
Price: $595.00
Circle 82 on Reader Service Card

LIMITER

• This limiter is named the "Gain Brain". It is actually two limiters in one; a high ratio peak limiter and an r.m.s. limiter with the attack and release times coordinated. The threshold of the r.m.s. limiter may be moved as much as 12 dB below the peak limiter threshold. It features a sequential light gain-reduction indicator, and peak and r.m.s. indicators. All lights are solid state and will never have to be replaced. The "Gain Brain" is the same physical configuration as the Kepex and will fit into all of the same hardware. A detailed bulletin is available.
Mfr: Allison Research
Price: $300.00
Circle 98 on Reader Service Card



STEREO CONSOLES

• The 212K-1 and 212L-1 are all solid-state audio control systems. The 212L-1 has eight stereo inputs, six of which will accept a mic, phono, or high-level input. Mixers seven and eight are wired for five remote lines. Impedances are easily selected by strapping. Both low level and high level inputs are available. The 212L-1 also has a mono output that will drive an a.m. transmitter while the stereo outputs are driving an f.m. multiplex transmitter. This model also has two stereo monitor output amplifiers plus a mono reverse cue amplifier. There are low-impedance stereo headphone outputs. Model 212K-1 is similar to above but has four stereo inputs with one, two, and three accepting mic, phono, or high level. Mixer four is wired for five stereo remote lines.
Mfr: Collins Radio Company
Circle 70 on Reader Service Card

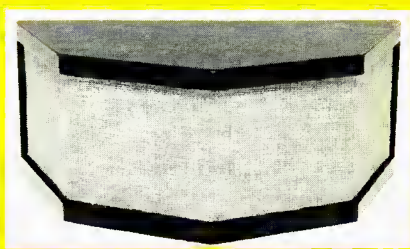
CONDENSER MICROPHONE

• Model C-37P is a variable-directivity condenser studio microphone. It is an improvement on this company's earlier C-37A. A low-noise f.e.t. amplifier is incorporated that provides a dynamic range of 130 dB. Noise level has been reduced by 10 dB and the overload point has been increased by 20 dB. In addition, the mic utilizes standard phantom powering, and may be used with any 48-volt phantom supply with 2.5 mA or greater capacity. Frequency response and directional characteristics remain unchanged from the older model. The mic is supplied with carrying case and microphone cover.
Mfr: Sony-Superscope
Price: $295.00
Circle 74 on Reader Service Card

WIRE MARKERS

• Self-laminating pressure-sensitive wire markers, with their built-in transparent shield to protect the legend, are available. A total of 450 self-laminating wire markers in a dispenser pack small enough to fit a shirt pocket can be conveniently carried by the technician. The markers are intended to service severe environments. They apply to the wire so that their transparent portion overwraps the legend portion, automatically providing protection. Abrasion won't obliterate the legend, nor will moisture, oil, solvents, or dirt loosen the marker. Three selections are offered. You can have numbers 0-9, numbers 1-45, and a combination of letters, numbers, and symbols.
Mfr: W.H. Brady Co.
Circle 72 on Reader Service Card

HIGH FREQUENCY SPEAKER SYSTEM



• The Microstatic is designed for use with AR and KLH speakers. It is claimed to have a 180-degree radiation pattern and flat frequency response over its range of 3.5 kHz to 25 kHz and thus gives the listener optimum high frequency response in any area of the room. It also will improve stereo localization of individual instruments and allows greater choice of speaker placement. Four direct-radiating surfaces are used across an arc of 135 degrees. The dispersion pattern of each driver overlaps the adjacent one creating a smooth transition and a uniform output across 180 degrees. Low distortion is achieved by the use of a special formula damping compound in the suspension of the radiating surfaces plus the use of a unique low mass diaphragm assembly.
Mfr: Micro/Acoustics Corp.
Price: $77.00
Circle 78 on Reader Service Card

(continued from page 16)

The contractor is eventually the one who becomes the devil or saint in this whole operation. We know, oh we know, what contractors will do.

the contractor is going to be responsible for doing everything in the specification. There will be no arguments. If you say you want an 8-inch block wall, and the drawing says an 8-inch block wall, he will put up an 8-inch block wall. That's all you have to do. So, *get a good architect.*

Now try to find yourself a good contractor. The contractor is eventually the one who becomes the devil or saint in this whole operation. We know, oh do we know, what contractors will do. Some can't even read drawings and some contractors don't even know what a drawing looks like. All their lives they have been told "put a wall up". In studios we don't say "put up a wall". We say put up a Type 1, 2, 3, or 4 wall. So *find a good contractor.*

There are all kinds: there is the guy who will say he can build you a studio at $20 a square foot. It will be the best in the world. You can reach 130 decibels; you won't hear a thing in the next room. Get rid of him quick–get him out. You may pay more for a good contractor in the beginning. You may pay $100,000 rather than $80,000. Pay the $100,000. The one who says $80,000 as a quote will cost you $40,000 more to correct what he should have done. You will never save money in getting a poor contractor or an incompetent contractor or a just plain stupid one. They are all around. If you can, get a list of three good contractors. We had to learn by our own experience. We had to teach those contractors who are now on my list.

The architect is an important guy. Either you will hang him on the wall or you will say that he is the greatest guy in the world. So, let's say you've got the architect as well as a beautiful space–uptown, west side, downtown, east side, wherever. The first step is to establish (this is the first time I am mentioning acoustics) the *acoustical design criteria.* This established our goals. What are we shooting for? What shall be the ambient noise levels? The air-conditioning noise? What shall be the intrusive noise from the building next door? What shall be the intrusive sounds from any kind of mechanical equipment, aircraft, especially sirens from fire engines? We can't wipe them out.

AMBIENT NOISE LEVELS

Generally, we mean the noise attributable to the air-conditioning system. You can take a big fan, install 25 silencers in, one after another, but put in a couple of diffusers and you have the flow producing 2,000 feet of velocity and you can't hear a thing. When we say ambient noise levels, we mean everything. We mean the fan noise and the aero-dynamic noise.

For studios, NC 20 can be used. That is very low. NC 25 is practical because to go from 25 to 20 will cost you a lot of money. So why pay for it when you don't need it? Particularly if you are doing a good rock group. You may get away with NC 60.

It is possible that you can be in a building where there is acoustical energy but the octave band level is very low. However, that beautiful mic that you paid $700 for is picking it up and your needle is going like mad. What you have to do is become a non-purist and start rolling off below 60 if you don't want to pick it up. You can get into a building with a lot of subway rumble and suddenly emerging in the building in which you spent a lot of money is 20 hertz.

In studios, an NC 25 is a very good number. We have never used NC 20 as a design goal. We have ended up with NC 20 but we did not cry about it. It is when you end up with NC 30 that we get concerned. NC 25 is a good goal. The smaller the space, the more significant the number because you can get into a very small speech studio with the diffusers three or four feet away from the mic. Under such conditions you may go for NC 20 but if you are doing a job and there is only one narration booth and you have got three of four studios at an NC 20, you are not hurting yourself. You could conceivably go down to NC 20 within a tight space for speech if there was a low ambient noise level.

Everybody must agree to what goal we are designing because the whole heating-ventilation-air conditioning system is going to be designed for this. The ducts will be sized for it, the acoustical liner selected for it. The diffusers, the ceiling grills, everything that is associated with the entire heating ventilation system, will be determined by this. If you go to a multi-zone unit, it is the best because you can get controls on your zones. All will be selected to eventually achieve this. After the studio has been build and in operation, and a system of balance established, then we can record. So we all agree that NC 25 is it!

ORIENTATION OF SPACES

Should the architect come to me and say "Bob, I have 8,000 feet here and we want to put an A and B studio and a mix down room, and a narration booth and an editing room. How do we put these all together?" Well, it really doesn't make any difference how we put them together but it does make a difference that we don't put a control room next to a mix down room, or a mix down room next to a studio. You just can't do it unless you are lucky to get involved with a building where you own the basement, then put them down there, the big ones. You can put a mix down room next to a studio because it is on grade. If you have to go up to the 30th floor of a high-rise building that has 3-inch concrete on a steel deck you are never going to get away with it. Forget it. What you need is separation. So you put the mix room down there and the studio up here. Get yourself some physical separation between your spaces.

ACOUSTICAL SEPARATION

By acoustical separation I don't mean sound transmission loss. I mean what would be the actual difference in sound level achieved between contiguous spaces. There is no correction for absorption at all; we don't care about that. All we want to know is if a guy generates 100 dB in one room, what are we going to measure in the other? That is

Get yourself some physical separation between your spaces.

You may be occupying a building where there is a lawyers' office downstairs. If you don't want to be harassed by telephone calls and a bothersome lawsuit . . . you will have to put in a floating floor . . .

what we are interested in. The next thing to decide is the acoustical separation we want. Now there are three types of studios that a guy can have; he can have a Cadillac, a Buick, or a Ford. Today if you go to a Cadillac studio—$60 a square foot—and you have 10,000 square feet, you are out of business before you start. We rarely do Cadillac studios today. We do Buick studios. We dislike doing Ford studios because when a guy gets a Ford studio, he says he should have taken a Buick, but it is too late.

What is a Cadillac studio? Everything is isolated, it is the old Johns Manville way of doing things. The floors, walls, and ceilings become isolated. Everything becomes very sophisticated. After you design someghing like this you begin to chew your fingernails because you wonder where you are going to get the craftsmen to build the studio. So a Buick studio is halfway between: you are getting good separation but you may not use floating floors.

In a strictly broadcasting studio and a d.j. studio in particular, you may not even need floating floors. And if you own the space above, you may not need isolated plaster ceilings. As a matter of fact, I'd be more concerned about the kind of sound the d.j. generates getting out of the studio than anything getting in.

If it is a recording studio, you have a different problem. You may be occupying a building where there is a lawyers' office downstairs. If you want to be harrassed by telephone calls and a bothersome lawsuit because your rock group is preventing them from making important decisions, you will have to go for a floating floor to avoid them. Put it in and spare yourself the aggravation.

The acoustical separation that is required is really dependent upon what kind of a structure you are living in. If it is a thin structure, a wood structure, etc. you can not put 8-inch thick concrete blocks on a wood floor structure unless the structural beams are fully capable. You don't find this, however. As a matter of fact, if you have a wood floor structure, I would not even recommend that you put a recording studio in there because the thing will just take off. You cannot effectively isolate it. If you get in some of these beautiful buildings that have 6 to 7-inch concrete slabs and reinforced beams, take it. Rent it. We can get good separation. If you go to New York's Electric Lady, we put studios side by side. We could only do that because we were on a slab on the ground.

The acoustical separation is predicted on the building you are in. Before you put your name down on that beautiful building's ten-year lease, you may find out that this structure will not permit you to have the acoustical separation you need. Now this is where we would come in (and the structural engineer would get involved with us) because we would say "Now, can I get a cavity wall here that weighs x lbs. a square foot?" He may say no. We tell the architect, who tells his client not to sign it. It is pointless to go ahead on a project knowing it is not going to work. You don't cross your fingers in this business; it is not a game. When a group invests money into a studio, and the door opens, everybody is rushing in wanting to get the thing moving. So don't play games.

What kind of separation do we need? If we start at 125 hertz at 50 dB and you zap up to 80 dB try to get it up to 90. You notice I did not say anything about 63 hertz. I don't have much data on what a building structure is going to do at 63 hertz, let alone what it will do at 31 1/2 hertz. The U.S. government is presently conducting field and lab tests on the response of a building structure to very low frequencies. When that comes out, we are going to be able to translate all the data into what it will be in the field with a lot of adjustments to it.

THE INFAMOUS MASS LAW

Believe it. It is true today as it was when it was first stated. I don't care what the various manufacturers say about their beautiful gypsum boards and partitions that have STC 60. That is all right if you are talking about one girl talking over here and the other talking over there. Normal conversation. Of course, that's good separation. But when you are talking about music, especially rock groups, you are talking about a tremendous amount of energy and these partitions just take off. It won't work—forget it.

For a broadcasting studio, yes. We did a studio in Los Angeles, one of the most unusual constructions I've ever done. Wood stud and gypsum board. Who would have believed it? But the studio's format allowed us to use gypsum board, but if they ever bring a rock group in, the whole roof will take off. We did get away with using gypsum, but only on that. You can also get away with gypsum board on speech studios. How much energy can you generate within a speech studio? Caruso may have hit 125 dB, but how much can you really get nowadays?

So acoustical separation cannot be decided until you know what kind of a building you are going to be in. Your building structure is your limit. If you are in a wood floor structure, you are never going to get over 65 dB. I don't care how much isolation you have.

ROOM ACOUSTICS

After we have got this whole structure made—all the partitions, windows, and doors—the architect asks "What

. . . if you have a wood floor structure, I would not even recommend that you put a recording studio in there . . .

We tell the architect who tells his client not to sign it.

shall be on the walls?" This is where we are in trouble. I know what I'd like to put on the wall but I don't know what the other guy wants. What kind of a sound are you looking for? Do you want a tight one, a bright sound, a sound with guts? My approach is I balance the room acoustically flat. Now we find out what particular sound is wanted.

We are not talking about the control room; the control room is a special thing itself. Some guys like it bright—I pull the highs down and bring up the lows. Acoustically—I am not even going to tell you about the parameters—everybody wants to do something different. "I want a bright sound for my strings," says one guy; another says, "keep the low frequencies down." So if we start out flat, we can do all of these things.

We do it with a number of elements—naturally we don't do it with a perforated partition because the holes are 1-inch on center and it doesn't absorb much sound on the high end. We do it with special acoustical treatment that can be put on the walls. On the ceiling, we do it with scattering elements. We may not necessarily splay the walls. If the room is small, a splay is in order. Principally, we are looking for scattering elements.

We don't believe in reverberation times. I don't think it means very much. I can show you two studios with the same reverberation time and one is great for recording and one is great for nothing. The way you can measure a reverberant time: I would say 0.8, you would say 0.6 or 1.2. You are talking 60 dB and I am talking the first 35 decay. What we really are saying we need to know is what is the ratio of the reverberant energy to the direct energy. That tells us a lot. If the ratio is very high, you have a very live room. (The direct sound energy is equal to the reverberant sound energy at 1 foot or 6-inches.) If it is a very dead room, the quality may be out at 8 feet, but at least you can start seeing some sensible relationship between a lot of studios and concert halls.

So we are really setting up in our own mind what to do: for lecture halls you take 5 or 3 (now I'm talking in numbers only), for studios you want to go down lower, perhaps to 2. If a guy says to me "This room is live," he may really be saying this room is live at the high end. And it really is dead at the low end. It depends on how the ear hears it. In studios we really are trying to satisfy two things: the response of a mic and the response of the ears of the musicians. they play there; they feel what is happening in the room. The room acoustics in the studio are different from the control room. There we are really concerned with hearing. What kind of loudspeakers are you using? What is the distance of the loudspeaker to the guy at the console? The control room is, I think, the most important room in the recording studio complex. If this room doesn't work, you can lock up the studio. We don't have to make the control room sound right to the guy who is doing the recording, all the mixing, all the work. And it can vary. I have asked numbers of engineers what constitutes a good sound in the control room and I cannot get a single opinion. I can get similar opinions on what is a good sound in the studio. In the control room they want a loudspeaker here, one here, one there.

So, don't play games.

Fortunately, we can communicate. We can take non-technical terms such as gut sound, tight sound, bright sound, and hard sound and translate them into a language of acoustics. I must translate the language for the architect, the general contractor, and myself.

SOUND BARRIER CONSTRUCTION

We have had our conversations with the owner, the architect is making his pretty drawings and writing good details. The next thing is to select our sound barrier construction. We have set the criteria. We have had a complete understanding and agreement with the owners that this criteria is what we are shooting for. If the owner says he doesn't care if he can hear pianos in the speech studio because he won't record in the studio when he is also using the music studio, so he doesn't have to worry about the acoustical separation. Great! I send him a telegram stating: "Pursuant to our discussion. . ." We put in the report that the piano will be audible in the speech studio. That is the best way, because when he reads what he said, he gets a chance to reconsider his statement. Once we start designing and punching in materials, once we have established the orientation of all these spaces you have got to pull the door shut and finish the design. During the progress of the drawing if you start expanding the studio, thinking it's made of rubber and it isn't, you will find you can't do it.

I know what I'd like to put on the walls, but I don't know what the other guy wants.

Since the room partitions represent the largest radiating surface in the horizontal direction, they are very important. The floor slab represents the largest ratio of surface in the vertical direction. So I am going to pin those together. We will keep the doors separate and then we will put in the windows and the ceiling. The ceiling could be a supplement to the floor structure, such as an isolated plaster ceiling or an isolated slab. Partitions obey the mass law. If possible, put in block masonry and plaster. It is acoustically sealed. Examine the block. In one job we took the block, wrapped in brown paper, to the Post Office and had it weighed. We were suspicious. It came through. You have to be a bit of a skeptic.

Be sure that when you use plaster that you are using sand aggregate. No lightweight material. Make sure it is sand. We are looking for mass. You can get more sophisticated construction but who is going to build it for you? There is no point of putting in a cavity partition which may use a 12-inch block with a 12-inch air space—no point in doing that on a 6-inch slab if you don't install an isolated floor because the sound energy in the slab will

We don't believe in reverberation time . . . I can show you two studios with the same reverberant time and one is great for recording and one is great for nothing.

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flank the block very efficiently. But there is a lot of sound energy that is going to go *into* that floor, go right *over* beneath the cavity block wall and you are going to wonder what happened. There is a limit to what you can do with that partition.

We have a rule of thumb. If you are going to be above 55 dB at the lower frequencies, you had better start isolating. Everything is just going to flank out on you and everything you erect is for naught. As an example: two studios used simultaneously and that are contiguous, you are wondering why you are getting contamination on your tape in studio A from music in Studio B. The whole reason is the sound flanking. It doesn't have to be the floor. We have seen a 12-inch block wall subdividing studios and then a 4-inch block wall as a common corridor and the guy said "What do I care about the music that goes out into the corridor partition, I am really concerned about the music in Studio A and B." But what he forgot about is the energy that enters into that common partition and travels along its structure and comes out into either studio A or B. So all of your enclosing partitions have to be selected so that you do retain the acoustical separation you want in the beginning. If you use a 4-inch block sometimes that block can be split to obtain a discontinuity. The separating partition cannot be unique and unrelated to the abutting partition. You could very well have an exterior wall with a column and not know it was a boxed in column. You erect a partition up to it and sound goes right through. You have to look at everything; we are just talking about the partitions, the adjacent partitions, the floor slab, and the floor slab above. If you are going into a building where the construction is uniform, you can say there is a 6-inch floor slab on the 6th floor, and there will be a 6-inch floor slab on the 7th floor. Check it out. Make sure. You can be surprised. But if it is the same, then you just have to worry about your flanking around partitions. More studios have failed because of flanking. Forget about sound leakage. If all my problems were sound leakage, it would be great because you can correct sound leakage problems.

Let's go to doors: it is our feeling that the only type of door that should be used in a studio is steel. First of all, it is dimensionally stable. It can be made with a great deal of precision. I have seen steel doors racked because they were dropped from the loading platform of the truck but I have never seen any warped. They are better as acoustical performing elements, far superior to wood. Don't use wood. Use steel. They will take a much greater abuse than wood. The only disadvantage of steel that I can think of is that it is difficult at a later date to put in a window. A vision port would be quite expensive. It would be better to call up the steel company and say "Send a new door with a vision port in it" than to have someone modify an existing one.

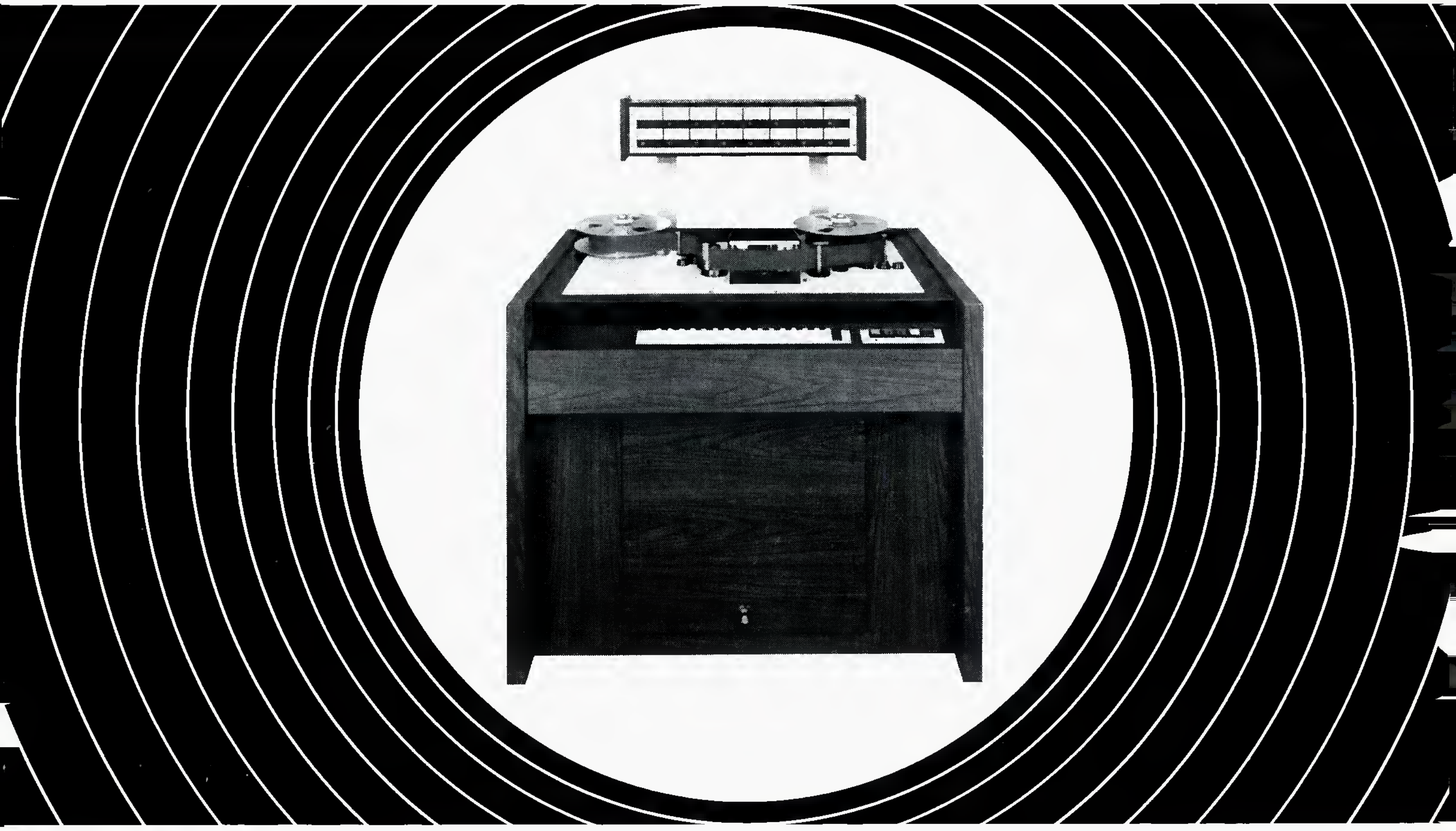
Of course, there is merit to it because there is a certain amount of discipline. When you order a steel door, you have got to make a decision then. Don't forget there are many types of steel doors. We are talking about sound insulating steel doors. They can be 4-inch thick to separate a t.v. studio from a storage room. Normally, they are 1 3/4-inches thick. There are sound insulating doors with acoustical performance ratings, some as high as STC 48, which we like. A 1 3/4-inch hollow metal standard steel door bought with pre-fabricated acoustical seals is good for STC 40. You can save yourself money. Don't go out and buy a steel door for $150.00 because it has an STC 40. Buy the hollow one. Put a drop seal on the bottom of the door and acoustical seals around it. You are going to get as much as a door rated STC 40. Remember that the ratings were obtained under the most perfect conditions imaginable. Five men may have been sent to the lab and they spent 36 hours preparing that door for its christening. You are not going to experience that with the carpenter on your job. He is just going to push the seals up and do it in one-half hour. It is a field condition problem. You must realize that. The STC 45 that you get in the lab will never be realized in the field. You may get 40.

WINDOWS

You can get any kind of window you want. You can get single pane, double, triple, ¼, 3/8-inch, acoustical glass. For recording studios we like acoustical glass. It has a very good response to the acoustical energy. We like to mix up acoustical glass with plate glass for reasons of economy, plus we feel they do a very good job. For speech studios we don't feel it is mandatory unless the criteria dictates we need separation between the speech studio and the control room of x dB. Then you may need acoustical glass. But, you must make every effort not to degrade the over-all acoustical performance of that partition. If that partition is rated for 70, don't put two pieces of 1/4-inch glass there. If this window represents 20 per cent of the wall area, it is a big element in that total wall area. You better be concerned. Get it up close to 70. You can get a difference because it doesn't represent the whole wall area.

Windows should be put in steel frames, properly grouted. The problem with wood is sound leakage. There are holes all over the construction. If you start out with steel, you are starting out with a good construction. Mount your glass in neoprene glazing channel. Again, if you can't find any glazing channel, there has to be expediency. When you select a substance as a substitute for what we are doing, you have to be sure that this substitute will not affect what you are trying to do. We substitute plaster skins for masonry only because we can not support a lentil above the window. The masonry above that might have been so heavy, we would have had to put additional structural steel in to support the load. It can be that we could get away with it. There are large panes of glass that can be obtained—but if you can't obtain the window, there is absolutely nothing wrong with butting the pieces of glass. I don't think you will appreciate a 3-inch wide divider in the middle of the window, particularly in a broadcasting studio where the console is close to the glass. You can butt the glass and seal the butt condition with silicone sealant that is clear. We like to butt our glass. The clients think it is really a panoramic sweep. You are not looking through posts. ■

If performance turns you on turn on the Scully 100



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The big secret? Like simple. Take the same Scully engineers that design the studio equipment that's been the standard of the business for years. Let them come up with the first really modular unit that lets you buy *only* what you need. Forget all the factory-loaded accessories and extras if you already have them on your consoles. Add them if (and when) you need them.

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What can you lose . . . except all your ideas about how big and how expensive professional recorders really are?

Dictaphone

Scully Division

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MELVIN C. SPRINKLE

Acoustics for Audio Men, part 2

In this part, the author covers the transmission and behavior of sound—both out of doors and in rooms.

Suppose we want to lay down a sound pressure level . . . how much power do we need and will the loudspeaker stand it?

NOW that we've had some of the basic physics underlying sound, let's consider something a bit more practical. Let's begin by considering something that (as far as I know) is only theoretical. (This is like r.f. transmission line articles that start out with an infinitely long line, which nobody ever built). I am referring to an isotropic sound source. Now, an isotropic source is a fancy and technically correct word for a sound source which is absolutely non-directional. Let's further stretch our imagination to position this isotropic source away up in the air and away from any reflecting surfaces, such as the ground. Let's also imagine that we are like *Superman* and able to fly up-up-and-away and approach this isotropic source without affecting its radiation pattern. Now since our source is radiating sound in a non-directional manner, the sound waves that emerge act as ever-expanding spheres of alternate compression and rarefaction of the ambient atmosphere. You will recall that in order to generate sound, power is required and this power is radiated from the source in the sound waves. We know that this is so because if we remove the power from a sound source, such as a loudspeaker, the sound output ceases—no energy was stored in the loudspeaker.

The surface area of a sphere is given by the following formula (snitched from a math text):

$$S = 4\,\pi\,r^2$$

where S is the surface in square units
$\pi = 3.1415$
r is the radius of the sphere

Mel Sprinkle has recently formed a new firm of consulting acoustical engineers—Sprinkle, Wildermuth & Associates, Kensington, Md.

In the last article we defined sound intensity as sound power per unit area. It therefore follows that since the sound from our isotropic source is being radiated in the form of an expanding sphere, the sound intensity may be expressed as the sound power in watts divided by the surface area of the sphere:

$$\text{Sound Intensity} = \frac{\text{sound power in watts}}{4\pi r^2}$$

From this formula it is seen that the sound intensity decreases with the square of the distance from the sound source (radius of sphere, r). This is the derivation of the famous inverse square law, which I might add, also applies to light energy.

Since sound intensity is not directly measurable, let's think about sound pressure level which is measurable. Knudsen[1] gives the following relationship between sound intensity and pressure:

$$I = \frac{P^2}{10^7 \text{ rho C}}$$

where I is the intensity in watts per cm^2
P is the sound pressure in dynes per cm^2 (microbars)
rho is the density of air in grams per cm^3
C is the speed of sound in centimeters per second

Knudsen points out that this relationship is not true very near the source, but we are far enough away that it does hold (say greater than ten wavelengths). This formula says that the intensity varies as the square of the pressure, or twisted around, that the pressure varies as the square root of the intensity. Since we saw above that the sound intensity dropped as the square of the distance from the sound source, then it follows that the sound pressure drops off as the square root of the distance squared, or inversely as the distance. In practical language, this means that if we measure a certain sound pressure at say 20 feet, then at twice the distance (forty feet) the sound pressure in a free field will be half of that at twenty feet. Since sound pressure is almost always measured in dB with respect to 0.0002 microbar, the sound pressure level will drop by 6 dB for doubling of the distance (sound pressure follows the *voltage dB* rule).

This relationship may be immediately applied to a practical problem. Suppose we want to lay down a sound pressure level of 80 dB at a distance of 100 feet from an *outdoor* loudspeaker, how much power do we need and will the loudspeaker stand it? Let's say that we are using an Altec Lansing 31A Horn and a 730C driver. Altec says that the driver will carry 75 watts, and that when using a warbled tone from 600 to 2400 Hz at a distance of 30 feet from the horn, the sound pressure is 104 dB with 75 watts (+48.8 dBm). One of the reasons for using Altec as an example is that they supply the engineer with quantitative data concerning the performance of their loudspeakers, as compared with some other manufacturers who say nothing or give unquantitative information dreamed up by Madison Avenue. With the above information, we are ready to construct a very interesting and informative graph. First, obtain a sheet of three cycle, semi-logarithmic graph paper (Keuffel & Esser #358-71 or equal). This can be procured from drafting or engineers' supply houses. On the log scale (abscissa) lay out distance in feet: 1, 10, 100, and 1000 as shown in *Figure 1*. On the ordinate, lay out sound pressure levels in dB using a scale which will cover the range of interest. Since the loudspeaker will deliver 104 dB at 30 feet, locate this spot and make a dot (point (A) on *Figure 1*). Then go to twice the distance (60 feet) and down by 6 dB, thus locating point (B). Then go to half the distance (15 feet) and up by 6 dB (110 dB) and locate point (C) (yes, two points determine a straight line but three points enable one to draw a straighter line). Then connect the points with a straight line, extrapolating to the edge of the paper. Label this *75 watts (+48.8 dBm)*. Next, on the 30 foot abscissa, count down by 8.8 dB and locate point (D). Also, locate points (E) and (F) which are 8.8 dB below points (B) and (C) respectively. Connect these points with a ruler, extrapolating to the paper edge. This curve should be labeled *10 watts (+40 dBm)*. We can then construct as many curves as desired, using the same technique. I have made a third curve at the one watt level (+30 dBm) which I find to be a handy reference. Our sound pressure objective being 80 dB (ref. 0.0002 microbar) at 100 feet is point (J) and it lies between the 1 watt and 10 watt curves. Specifically it is found to be 5 dB above 1 watt or +35 dBm or 3.162 watts. While the 3.162 watts will deliver the desired sound level, I'd use a minimum of a 30 watt amplifier into this speaker because this will give 10 dB headroom. Headroom refers to the peak-to-average ratio in program material and good engineering practice requires 10 dB or 10 times the power. The graphical method described above does, however, give some quantitative information on power requirements but without consideration of headroom.

Since sound intensity is not directly measurable, let's think about sound pressure level which is measurable.

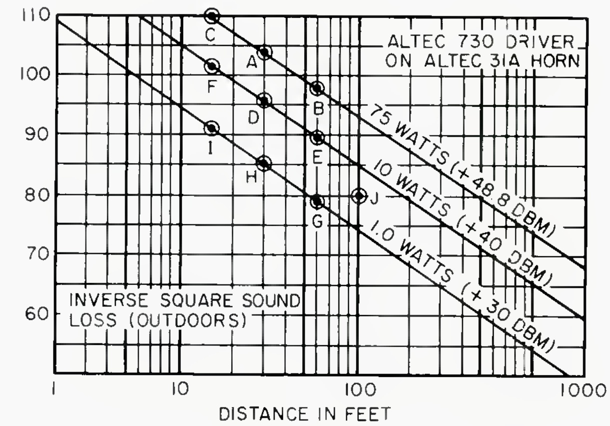

Figure 1. An Altec 730 driver on a 31A horn. This is inverse square sound loss outdoors.

Now that you've made your curve for the speaker, you should file it along with other engineering material for the next job. Curves of this type can be constructed for any loudspeaker whose manufacturer gives quantitative data on its acoustical performance. The essential information is: so many dB sound pressure at a specified distance with a specified electrical power input.

Several words of warning: 1. This method applies *only to the outdoor case*. There are other rules for indoor loudspeakers, which we'll come to in due time. 2. The acoustical data from the manufacturer should use a warbled tone or pink noise. Single frequency tones are *valid for that frequency only*.

The reader may have noticed that we used a rule which was not specifically pointed out until now: this is that there is a 1 to 1 linear relationship between sound pressure in dB (ref. 0.0002 microbar) at a fixed distance to the electrical power in dB. If you want 3 dB more sound level, then dump 3 dB (twice) the power into the loudspeaker. This may be done until after adding the 10 dB for headroom the manufacturer's power rating of the loudspeaker is reached. Never but never exceed the power rating with the headroom added.

We might also mention that the 6 dB loss in sound pressure with doubling of distance rule follows the curve of an equilateral hyperbola (like capacative reactance). The sound pressure falls off rapidly for short distances, but drops more slowly at greater distances. For example, if we go from 10 feet to 20 feet we lose 6 dB; we also lose 6 dB in going from 500 to 1000 feet (naturally the sound level is less but the change is still 6 dB). This can be seen from *Figure 1*, for with 10 watts we get +78.5 dB at 200 feet and +72.5 at 400 feet. This is the reason that outdoor sound systems are heard at remarkable distances in a quiet ambient noise level.

The calculations above neglect reflections from the ground, nearby buildings, etc. These do affect the results. For example, sound is absorbed by grass and snow, but reflected quite well from concrete or asphalt roads. Sometimes these reflections can be helpful and sometimes harmful. There are no rules; each location should be analyzed for potential reflecting surfaces. The reflections, we might add, assist the carrying power of a gasoline lawn mower noise on a quiet Sunday morning!

Now that we've covered the great out of doors, let's go inside and talk about indoor acoustics.

If we consider a church where the minister is in his pulpit giving us the word on Hell, we note that the sound from his voice, without amplification, is louder that it would be outside, also without amplification. This is because the room has walls, floor, and ceiling which confine the sound, not allowing it to spread indefinitely as it would outside. Sinçe the production and maintenance of sound requires power, which is radiated by the source, a source of steady sound such as a hissing radiator, or an opera singer hanging on to a long passage will build up in intensity until the sound losses equal the sound production, or said another way, an equilibrium condition is set up. The sound losses occur because of leakage through walls or doors or windows to the outside or to other parts of the building, or by direct absorption in certain materials which absorb sound and convert the energy to heat. When the steady sound is suddenly terminated, the sound will persist until it decays away; this effect is called reverberation and we will consider it at some length later. The buildup in sound by a steady source until equilibrium is attained is called the *room gain* and was treated in a classic paper by Hopkins & Stryker[2]. Room gain is sometimes beneficial as well as sometimes a problem.

Returning to our minister, we note that persons in the congregation will hear his voice by the direct transmission of sound from the talker to the listener. The mechanism of transmission in this case is identical to the outdoor case, and the sound pressure level from the direct transmission component will decrease by 6 dB for every doubling of the distance (inverse square law) just like outdoors.

But since the room has walls and a floor and a ceiling, and sound emanates from the talker in many directions (though not necessarily uniformly in all directions), it travels to the surfaces of the room where it is reflected. It then proceeds until it either reaches the listener or another surface where it is again reflected.

Thus our parishioner in the congregation hears the minister's voice by two routes: the direct transmission component mentioned above, and a component which has undergone one or more reflections from the room surfaces. This component is called the *reverberant* component, and since it has undergone who know how many reflections it is random in nature. It is also delayed in time, for it has traversed a longer path between the talker and listener and therefore has required a longer time to get there. The reverberant component may be nearly as loud as the direct component, especially if the room is said to be reverberant—having walls, floor and ceiling which are good sound reflectors and poor absorbers. If the reverberant component is almost as loud as the direct component, then our parishioner has difficulty in understanding the words because he hears them once by direct transmission, and then one or more times later and delayed in time. This blurring effect occurs even though the time delay is not long enough to be considered as a true echo.

The reflection of sound from a surface, mentioned above, takes place in the same manner as in an optical mirror, or a billiard ball hitting the resiliant bumper at the edge of the table and taking off in a new direction. The reflection of sound follows the same laws as light and billiard balls, in that the angle of incidence equals the angle of reflection, both being measured with respect to a normal or line perpendicular to the surface at the point of impact. It follows that the net effect is the same as if there were an acoustical image behind the wall and radiating the reflected sound. This is the same as optical reflection wherein a virtual image is created in a mirror. Architects and acoustical engineers often use models of auditoriums illuminated by rays of light (passed through a comb) to locate potential troubles in reflected sound or focusing effects produced by curved surfaces. This is known as *optical acoustics*!

Room gain is sometimes beneficial as well as sometimes a problem.

When a sound wave encounters a surface, a portion is reflected while the remaining portion is absorbed, with an accompanying loss of energy. The energy is, of course, not lost (law of conservation of energy) and appears as heat. But don't worry, the power in sound waves is so small that there hasn't been a building set on fire due to sound waves being absorbed, even from a politician's oration. Some materials such as acoustical tiles, carpet, glass wool, or rock wool are efficient absorbers, while other materials such as marble, water surfaces, concrete, etc. are very poor absorbers. Acoustical scientists rate materials by what is called an *absorption coefficient* which is a number having values from 0.01 for marble or water to 0.8 or 0.9 for acoustical tiles. A material which completely absorbed all sound would be rated as 1.0. The coefficient of absorption is really a percentage and indicates the percent of sound energy which is absorbed per reflection. The coefficients are also functions of frequency, since most materials are much better absorbers (have higher coefficients) at higher frequencies than at low frequencies. The coefficients may be obtained from handbooks, or from text books on acoustics, and for specific brand materials are published by the manufacturer of the product. The coefficients depend also upon the method of installation of the material, and the Acoustical Materials Association has standardized on the several methods of mounting. An acoustical tile may have a coefficient of 0.7 when installed as a hung ceiling (#7 mount) and 0.4 at the same frequency when nailed to furring strips, (#2 mounting). Values given by members of the Acoustical Materials Association based upon tests at Riverbank Laboratories or other reputable testing facility may be relied upon.

If one knows the absorption coefficients of the materials which cover all of the room surfaces, then the total absorption may be easily calculated. Absorption is measured in units called *Sabines*, a word derived from the name of Wallace Sabine who was the father of modern acoustics as an engineering discipline. Sabines have the mathematical dimensions of square units, usually square feet. For example, if our church has a carpeted aisle with a coefficient of 0.69 and has an area of 100 square feet, then the absorption of the carpet is: 0.69 x 100 = 69 sabines. We might add that an open window with an area of 1 square foot has an absorption of 1 sabine for frequencies whose wavelength is comparable to or less than one foot.

During the time that a sound wave is traveling from one reflecting surface to another, it undergoes inverse square law loss, so that there is considerable attenuation after several reflections from absorptive surfaces.

We must also add the comment that the audience itself is a good absorber of sound. Tables of absorption in acoustics texts usually give these values. For example, an empty church pew has an absorption at 500 Hz of 0.40 sabines per 18 inch length without cushion. Cushioned pews are rated at 1.8 to 2.3 sabines per 18 inch length. A seated audience runs between 4.0 and 5 sabines per person depending upon spacing. The audience must be considered in auditorium acoustics.

By Murphy's Law, the usual case is that a hall has excessive reverberation and the owner turns to a sound engineer for a sound system.

If an opera singer hits high C and holds the note so that a continuous sound is produced, and suddenly terminates, the direct sound that reaches the listener will also cease as soon as the last sound energy reaches him. The sound waves which are being reflected about the room will, however, continue to travel until their energy is dissipated. As these pass the listener's ear, he will continue to hear them as they gradually die out or decay. This prolongation of sound is called *reverberation*. If the sound in a room dies out quickly the room is said to be dead as a broadcast studio; if very slowly as in a cavern or large cathedral, the room is said to be live or highly reverberant.

Reverberation is very important in the acoustical design of speech rooms and auditoriums and has a profound effect upon the comprehension of speech as well as the enjoyment of music. In a highly reverberant or live room, speech is difficult to understand at some distance from the talker, because the persistance of the reverberant sound reaching the listener blurs or confuses (because of the overlapping of syllables or even words). Organ music benefits from a high reverberation as we all know, as does choral music such as hymn singing by the congregation.

The amount of reverberation is measured by what is called the *reverberation time*. This is the time in seconds which is required for a steady sound level to decay (or lose sound pressure level) by 60 dB (decrease to 1/1,000,000 of the value when the source was suddenly terminated).

Since reverberation in proper amounts can be beneficial, there have been established values of reverberation which are considered as optimum for a given set of conditions. These may be found in textbooks on acoustics and are determined by two factors: the volume of the room in cubic feet, and the general nature of the use to which the room will be put, *e.g.* cathedral, general purpose auditorium. small church, lecture room, etc.

By Murphy's Law, the usual case is that a hall has excessive reverberation and the owner turns to a sound engineer for a sound system. Now we must point out that the installation of a sound system *cannot* correct bad acoustics; but it is true that with present day knowledge of sound system design, a sound reinforcement system can be designed which will enable the audience to hear well, even with bad acoustics. Indeed, it is possible, as has been reported in the literature[3,4], that as long as the talker uses the sound system he can be heard well, and with complete understanding, but the instant that he departs from the microphone the sound is confused and muddled. We must also mention that in those "acoustically difficult" cases, the sound system designer must not be hampered by restraints from non-technical people, such as: "we must not see the loudspeakers" or "you *must* put the loudspeakers here," etc. If such restraints are laid down, the sound man may be well advised to walk away from the job and let the lay experts install the system. There are rules of Nature which must be followed, regardless of the whims or opinions of building committees. On the other hand, if an engineering design is prepared and properly presented, together with adequate knowledge of the laws of Nature, a man who knows his stuff can sometimes convince a building committee. Furthermore, we have *never* said that loudspeakers cannot be concealed or installed in an attractive enclosure as long as the enclosure and the location are acoustically correct. *Reference 4* gives an example of such an installation.

In the next installment (due to appear in this magazine next month) we will make an acoustic analysis of a typical conference and briefing room, and show how such an analysis can be used to determine the proper location of the components of a sound reinforcing system. ■

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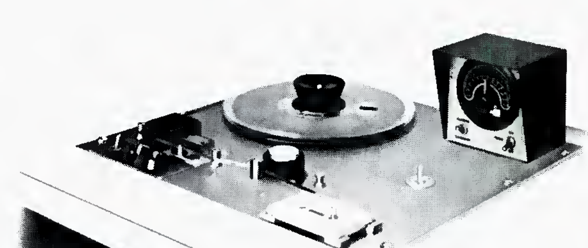

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The Audio Engineer's Handbook

GEORGE ALEXANDROVICH

EQUALIZATION IN SOUND REPRODUCTION AND REINFORCEMENT SYSTEMS.

• Until recently it was up to an acoustical engineer to apply acoustical treatment to a studio, theater, or auditorium and to select microphones and speakers which would acoustically match the response of the environment. Lately the acoustical deficiencies of many rooms are being compensated by electronically equalizing the frequency response of the amplifying system. Equalization accomplishes almost everything that acoustical treatment does except decreasing reverberation of the room.

The idea of equalization is not new. In the professional sound-recording field, equalizers have been used for years to compensate for the varying characteristics of microphones, room (studio) acoustics, and also to reduce the inter-mic crosstalk by limiting the bandwidth of the individual mic input circuit. In home-entertainment equipment the loudness contour and tone controls help accomplish this. One of the main reasons sound-reinforcement equalization thasn't been extensively practiced before was a lack of the right equipment. Several manufacturers finally jumped on the bandwagon and a variety of variable environmental equalizers with high Q and narrow bandwidth have become available.

Because of the relative precision and knowledge of what has to be done and how equalizers have to be set, manufacturers prefer to distribute the equipment through their dealers who are trained in survey installation and adjustment of the equalizers. They are equipped with automatic plotting devices in order to speed up and simplify their job.

Equalization of a system consists of acoustically obtaining the response of the system and then using a full complement of filters for equalizing until response is reasonably flat. At least one manufacturer, after this is done, then sells the customer only the filters he needs to accomplish this. Other manufacturers sell a package consisting of which cover the entire equalizers range. Even if you need just one filter you have to buy them all. On the other hand if the acoustics of the auditorium or room change you are equipped to cope with them. Another system offers filters which can be tuned over a limited range of frequencies. This method offers higher precision in tuning out individual sharp peaks but accomplishment requires the services of a better equipped and more experienced specialist.

All of these methods are more than satisfactory and all of them, executed properly, can enhance the performance of a system twofold at least. Aside from the fact that a correctly equalized monitor system would reproduce sounds more faithfully, sound reinforcement systems, can be set for much higher reproduce levels without feedback. However there are scores of systems and installations with much less critical applications and they don't justify the high cost of professional equalization. The purpose of this month's column is to work out simple methods of charting your own frequency-response characteristics of the installation and determine what has to be done to make the system sound and perform better. In future columns the design of simple filters and equalization techniques will be described.

As you know, most speakers and microphones are designed and tested in anechoic chambers. They are designed to work well within an environ-

The Shure SM58 *self-windscreened unidirectional* microphone is ideal for broadcast uses such as remote news, sports, interview and vocal recordings because it eliminates or minimizes the irritating "pop" caused by explosive breath sounds. With the SM58 you will have the peace-of-mind assurance that you're delivering the quality audio that goes with pop-free pickup. It's great for studio announcing, too—or wherever the announcer or vocalist has the audio-degrading habit of "mouthing" the microphone. Of course, the same filters that eliminate pop also do away with the necessity for an add-on windscreen in outdoor uses.

On the other hand, the unusually effective unidirectional cardioid pickup pattern (uniform at *all* frequencies, in *all* planes) means that it is a real problem-solver where background noise is high or where the microphone must be operated at some distance from the performer. Incidentally, but very important, the SM58 tends to control the low frequency "boominess" that is usually accented by close-up microphones.

All in all, close up or at a distance, the Shure SM58 solves the kind of ever-present perplexing problems the audio engineer may have felt were necessary evils. The SM58 might well be the finest all-purpose hand-held microphone in manufacture today. And, all things considered, it is moderate in cost.

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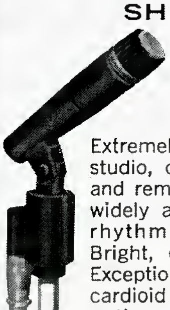

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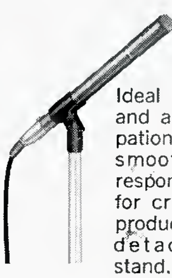

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NEW IM ANALYZER
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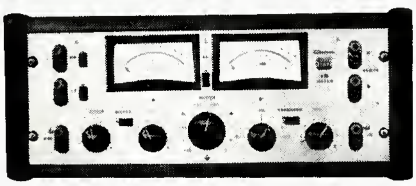
Extended measuring range and high-speed readings are the outstanding features of a unique new Intermodulation Distortion Analyzer introduced by Crown International recently. The American firm is known for its line of Crown precision professional tape recorders.

This analyzer was developed to meet the production line requirements of the Crown DC300 lab standard amplifier. The first need was for accurate measuring capability through 0.01%. This analyzer guarantees a residual IM level of less than 0.005%, with seven full-scale ranges from 100% to 0.1%.



The second requirement was for an instrument simple enough to be operated by production-line personnel and rapid enough to make sequential readings across the entire power band. The Crown analyzer meets the challenge by reducing measuring time from minutes per reading to just seconds. This is accomplished by a "tracking" function, using two meters and a ganged input/output gain control. The input level is set using the calibrate meter, and distortion is immediately read on the percentage distortion meter. Successive readings at 5db increments take under five seconds each. The entire operation is completed in less than one minute.

Solid state construction, utilizing FETs, makes the Crown analyzer highly stable and uniquely compact, measuring 7x19x7 inches. Rack mount list price is $595. Write for spec sheet to CROWN, Dept. DB-1, Box 1000, Elkhart, Indiana, 46514.

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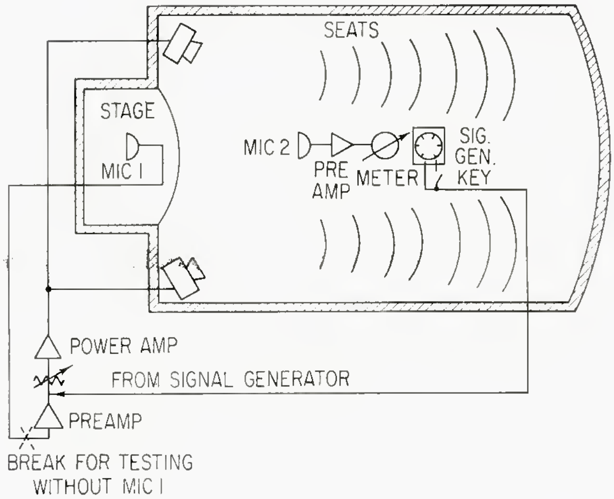


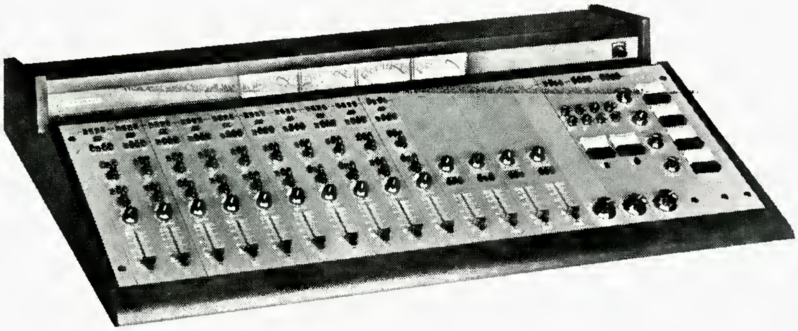
Figure 1. A typical test setup for measuring frequency response of a theater system.

ment which is passive and doesn't influence the performance of the transducers (speakers and microphones). Heavily-damped speakers are less influenced by the acoustical qualities of the room while high-efficiency types are more affected. But with today's extremely low output impedance power amplifiers, electrical damping of the speaker makes it less susceptible to external influences. Mechanical coupling of the driving coil and the speaker cone still remains the determining factor. Some reflected frequencies from the walls of the room return to the speaker and beat with vibrations of the speaker cone.

Those are the factors that make speakers sound differently when they are set in a room and listened to. In a simple and sound reproducing system for pre-recorded information, equalization serves the purpose of helping to reproduce the sound more faithfully to the original and serves no other purpose. In sound-reinforcement systems, aside from equalizing for better sound, we are seeking higher acoustic operating levels. If the system reproduces a certain frequency more strongly than others this frequency will be picked up by the mic sooner and stronger and will therefore be amplified more, sometimes enough to start acoustical feedback. If we remove all those troublesome peaks we can increase the maximum operating level of the system (sometimes) as much as 10 dB. Considering that 6 dB of voltage gain is twice the power, this gain in loudness can be substantial.

How do we measure the response of the system and draw a usable curve? First of all we need test instruments. If sound-level meter is available it will replace the otherwise needed calibrated microphone, preamplifier and VTVM We also need a variable frequency signal generator and a push-button momentary switch or key. This is needed to send the generator signals for short periods since long-duration signals at high levels are not only objectionable to the person who is measuring the response, but can also damage the speakers. Most speakers are rated for music power but not for r.m.s. tones. FIGURE 1 shows the suggested location of test gear for a typical theater-type job. It is important that the response of the mic used for measurements if a sound-level meter is not at hand) be known as accurately as possible. Even if the mic is not flat a derived curve can be corrected for the mic response deficiency. Place the mic where most critical listening will be done. In a control room place the microphone where the mixing engineer sits. Just as in your recreation room you would like the sound to be best in your favorite arm chair, so in the theater you aim for the best sound in the favorite middle front of the orchestra.) Make sure that stage mics and speakers are installed permanently and that they will not be disturbed after the tests, since a change in location of the speakers and mics will change the response of the system. This is not so much because of a change of the basic response of the speakers dut due to the ratio of the direct sound we hear to the reflected waves which bounce off the walls and come back.

Once the setup is completed (without sending any signals through the system) measure the ambient noise of the room or hall. Reproduced sounds from the speakers should register on the meter at least several dB higher so that the obtained results will be as accurate as possible a (because high ambient noise may mask sharp dips in response). In a sound-reinforcement system take two readings—one with the microphone open and one with it closed. The difference in results will serve as an indication of the effect mic-to-speaker coupling has on the response of the system. The more pronounced the effect, the higher the coupling and the lower the reproduced levels can be expected from the setup. If a sound meter is used, set it to the flat position. A VTVM has



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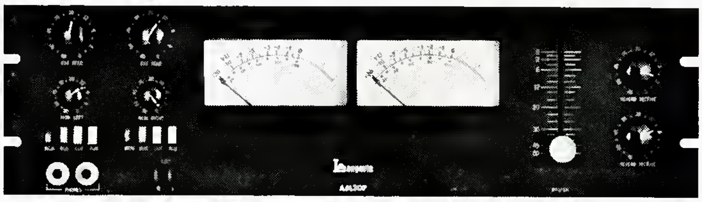

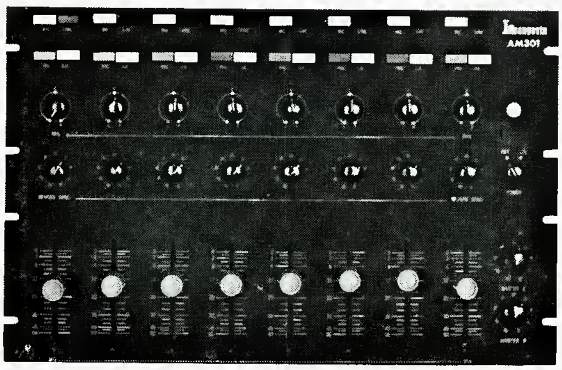

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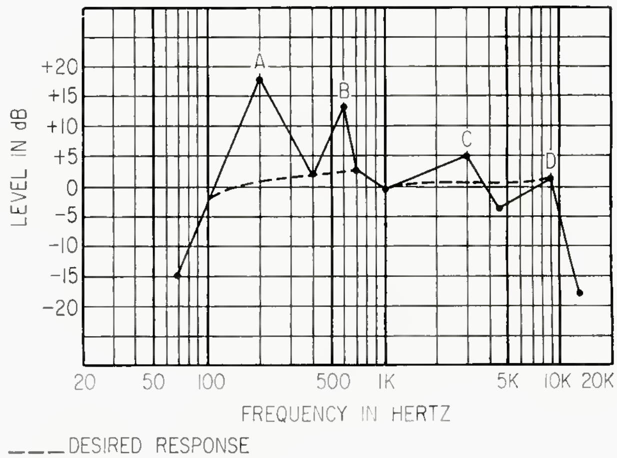

Figure 2. An example of a frequency response plot of a system, showing high and low points.

to be least sensitive enough to measure levels as low as −60dB otherwise an additional preamplifier has to be used after the mic. Sometimes 1:10 step up transformer will do the job if the system lacks 20 dB or so in sensitivity.

Adjust the signal generator to 1000 Hz. Adjust the output of the generator until the meter registers an increase in level over the ambient noise of at least 10 dB. Observe a 'scope for a clean sine wave that is clearly distinguishable from the noise signals. If the wave form is not clear, raise the level further until it is fairly clear. From this point on log all meter readings. Consider the level shown on the meter from 1000 Hz to be the reference level and mark it 0 dB. All other readings above 1000 Hz will carry a positive sign and ones below will be negative. Depress the key at the signal generator and start changing frequency upwards. While you change the frequency observe the meter. All fluctuation of less than 2 dB can be disregarded. Keep increasing the frequency until the meter registers a well-defined peak or dip (several dB) in the frequency response. As the meter goes through the maximum deviation, note the frequency and magnitude. Log the result. Continue changing the frequency until you reach a second peak or a dip—and so on. Don't be surprised if you find peaks 10-15 dB above the average level. After reaching the upper limit of the audio spectrum return the signal generator back to 1 kHz and continue changing the frequency downward. Keep signal going through only long enough to achieve the reading. With a little practice a few seconds should be plenty of time to get a good reading. When all frequencies are covered and all readings are marked down on paper it is time to take either the frequency-response paper (linear Y or vertical graphing, periodic log for X or horizontal divisions)—or prepare your own grid paper. It is really important to have a visual presentation of the system's response, so that you know whether a particular point on the graph is a dip or peak and which peak is more important to equalize out than the other. Figure 2 shows such a typical graph. Peak A is the highest and should be removed first, peak B is almost as high as peak A and can cause as much problems. Peak C although it measures 5 dB above the 1 kHz level may not be as objectionable because if all frequencies below peak C are equalized to the level of 2-3 dB above the 0 level then the deviation of peak C is only 3 dB or so. Sharp rolloff in the sample graph of the high frequencies can probably be explained by poor speaker construction or absorption characteristics of the room. Also, directionality of high-frequency waves should be taken into account. Speakers may not be pointed directly at the test mic and you may not be measuring much of direct sound from the speaker but may hear considerable amounts of reflections. Although this month we only covered the method of plotting frefiuency response, it would be advisable to be prepared to perform some sort of equalization once you are set up for testing.

Quite a few systems can be improved considerably by just removing a single peak, others may need a simple band pass or band stop filters to improve performance. Whatever we do to a system we must remember that even the most thorough efforts may not bring the completely desired degree of improvement because the system performs differently with an audience. Also if you listen from a different place in the house than the one where response measurements were taken, results may be different. There are other factors creeping in we shouldn't forget about such as phase cancellation and additional effects.

Without the trying to discourage the reader, I am stating the fact that the technique of system equalization is a big step in the right direction which eventually will be amplified by other improvements in technology and equipment.

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The Sync Track

JOHN WORAM

THE CASE FOR HUMAN ENGINEERING

Herewith, we begin a new monthly column to be devoted to the special provinces of the recording industry. Its author is no stranger to this field. He is a recording engineer based with RCA Victor in New York and, as such, has put in more than his share of long days and nights at the console.

Mr. Woram promises us that he will be responsive to reader inquiry and request on any subject pertinent to the field. That which he does not himself know the answer for, will be properly researched to give you the information you need. So look at this new effort in the same light as our other columnists, all of whom are ready to feed back inputs you may place. Ed.

● There is apt to be some amount of mental pressure involved in sitting behind a mixing console during an important live session. On a modern console, there may be as many as 600 variables (mixers, echo selectors, equalization, *et al*) any of which may need adjustment during a take. As the number of controls increase, so does the probability that you will sooner or later adjust the wrong one, and maybe blow a take. Very embarrassing. The musicians make funny remarks, and the producer makes remarks that are not so funny.

But what if a little mistake meant disaster rather than embarrassment? Suppose you were at a missile site and you pushed the rocket-launch button because you mistook it for a cigarette lighter, or, on the way into lunar orbit you fired a left retro when you really meant the right one? Obviously, little goofs like these must never occur. Once you've blown up Washington or crashed into the dark side, you can't apologize and ask for a re-take. There's got to be some way of stopping mistakes before they happen, and of course, there is. It involves, among other things, intensive training, a good night's sleep, and a confusion-free set of instruments in front of you. All of these little amenities have been built in to such critical programs as missiles and space probes.

Meanwhile, back at the recording studio, the training may not have been so intensive, and you probably haven't had a good night's sleep all week. And to top things off, there may be much confusion built into your console.

There may not be much that can be done about the training, or getting to bed on time, but there certainly is something to be done about the console. It's called *human engineering*. It involves not so much the electrical, as the *physical layout* of a piece of equipment. It means designing equipment without built in confusion. For space science, it helps keep astronauts alive. For recording science, it helps make life just a little easier.

For example, most recording consoles are of modular design. With minor variations, a module will contain: preamp pad, fader, preview, equalization, muting, and so on. Controls are usually all placed above the slide fader, and are designed to look practically identical. More often than not, a row of four or more round black knobs are found on each module. And it looks very impressive—until you get 24 modules in a row. It then looks very confusing.

For instance, in the middle of a take, the producer has a vision—"more lows on the bones", he says. The trombones' microphone was plugged into position 13, so you quickly count over to 13 and tnen up to knob 3, which is the low-end equalizer. If you haven't arrived too late, you twist the knob, and *voila!*—nothing happens. Why should anything happen, the eq. switch is in the eq. out position? As soon as you catch this, you quickly punch it in, and the lows come riding in on a sonic tidal wave. Surprise, surprise! It seems the low end was already up a few notches from flat before you cranked it up. But you didn't notice it because the knob is round and black, and the marker is on the upper part, well hidden from your eye by the knob's bulk.

Preposterous? What do you think? After a few hours staring into the lap of a console, the knobs have a way of staring back at you, and being where they shouldn't be when you want them.

This is where *human engineering* comes to the rescue. To over-simplify, human engineering says that all knobs don't have to be round, black and shiny. The military spends a fortune designing equipment that is, in addition to being electronically A-O.K., as human proof as is possible. One prominent military contractor has spent years analysing the behavior of equipment operators. The idea is, that by studying the relationship between man and machine, the machine can eventually be refined into a device that is almost impossible to mis-operate.

Much of this human engineering technology is available to the recording industry practically for the asking. A lot of it is, to be sure, irrelevant or impractical. But there is a nucleus of information that would be worth investigating by anyone planning a new construction project. In any man/machine situation, there must be a two-way communication system. The operator communicates with the machine—in our case, a console—by manipulating the various controls. The console communicates via its visual read-outs, which may be colored lights, switch positions, patch cords, etc. The more obscure this feed back is, the more you will be apt to make errors.

One obvious example of human engineering is the *tactile shape* control knob. That is, a knob with physical characteristics that indicate its function. Thus, at the missile site, the *self destruct* button is square, and the *light dimmer* is oval. Now, if you want to raise the lights a bit and you can feel corners, you know you've got your hand on a no-no. Reducing this to our needs, a square *echo-send* knob will not be confused for an oval *equalization knob*.

To design the ultimate console in terms of human engineering is no slight task. The designer must be able to separate the essential reliability requirements from the variations suggested by the personal tastes of various operators. Think about the little built-in stumbling blocks you may encounter in your work, whether it's behind a mixing console, or at a lacquer cutting lathe, or, wherever. Jot down your thoughts and send them to this column, along with any proposals you may have for improvements. We'll try to process the information received, and with a little help from our friends in the human-engineering game, perhaps we can come up with some ideas for a more confusion-free way of doing things.

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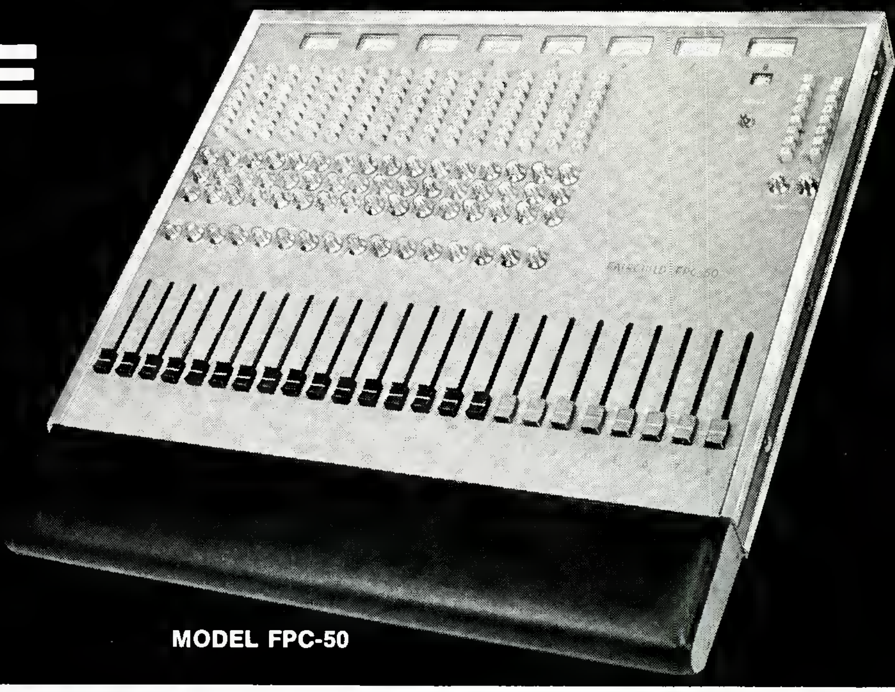

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The Feedback Loop

ARNOLD SCHWARTZ

● The presence of two major all-news radio stations in the New York City metropolitan area attests to the popularity of this format. Back in August we took a look at one of these stations, WINS. This month we will talk about the other all-news station, WCBS, located on the Avenue of Americas and 52nd street in New York City. While WINS' studios were designed to be an all-news operation, the present WCBS installation was originally intended for a less specialized format. Nevertheless, the original installation, completed in 1965, was designed with sufficient flexibility to allow the changeover to be made with relatively few changes and additions.

The previous WCBS format was one where much of the programming (talk shows, panel shows and music) originated in the studios. In the present format, news and other information is constantly flowing from outside sources into the studio complex. Typically, incoming news is recorded in production studios, and the edited program is then packaged on a cartridge tape to be utilized in the on-air studio. When the change came it became necessary to have a large number of incoming lines available. Accordingly, lines for the incoming feeds were installed, and fourteen-position selector switches were located at control consoles and tape recorders to enable selection of the feeds at these locations.

Now, virtually all broadcasting originates in control room B (FIGURE 1) which has a ten-channel McCurdy console. To accommodate the frequent live newscasts, involving two or more reporters, there are four microphone inputs from studio B. Studio B is shown in FIGURE 2, with control room B seen through the window. The largest of the three cabinets on the desk (directly in front of the announcer, Jim Harper) houses a selector switch which makes incoming lines available for monitor purposes. As can be seen from the floor plan, studio B is large and easily accommodates four people. The three remaining control rooms are used as production areas. In addition there are two edit rooms. Control room A, adjacent to B, (see FIGURE 1), is also the central control where jack fields, transmission equipment, switchers, and f.m. automation are located. FIGURE 3 is a photograph of control room A showing the logging recorders (left), and the f.m. automation equipment.

Originally all four control rooms alternated as on-air studios, and a relatively sophisticated master switcher was installed to transfer the outgoing program line from one studio to another on cue. The engineer in the upcoming control room punches a preselect button mounted on his console, and the outgoing program line will then be transferred to that console when the engineer in the on-air control room actuates the release button at the end of his program. The master switcher has eight inputs, and four outputs; *a.m.*, *f.m.*, *network*, and *spare*. An overide in the central control room gives instant access to the four outgoing lines. In the present format the master switcher is not as active as originally planned, but it does provide a very convenient method of switching studios.

The CBS Netalert system is a vital part of the WCBS operation. Netalert is a two-tone signalling system which is sent over program transmission lines simultaneously with the program signal itself. The transmitted code actuates a receiving device that monitors the network program line, but its presence on the line does not disturb the listener. Netalert signals originate at the CBS network facilities in New York, Washington D.C., and KNX in Los Angeles. The code is initiated by dialing a number from one to nine which then sends out one of the nine available messages. Each of the CBS affiliates is supplied with a receiver that decodes the tones. Remote indicators operated by the receiver unit are installed at control consoles and other strategic locations. The receiver output may be used to automatically initiate a sequence, or used as a cue signal to the engineer.

Here are some examples of how the Netalert system is used. Code number *one* signals a cutaway cue for local commercials during the hourly network

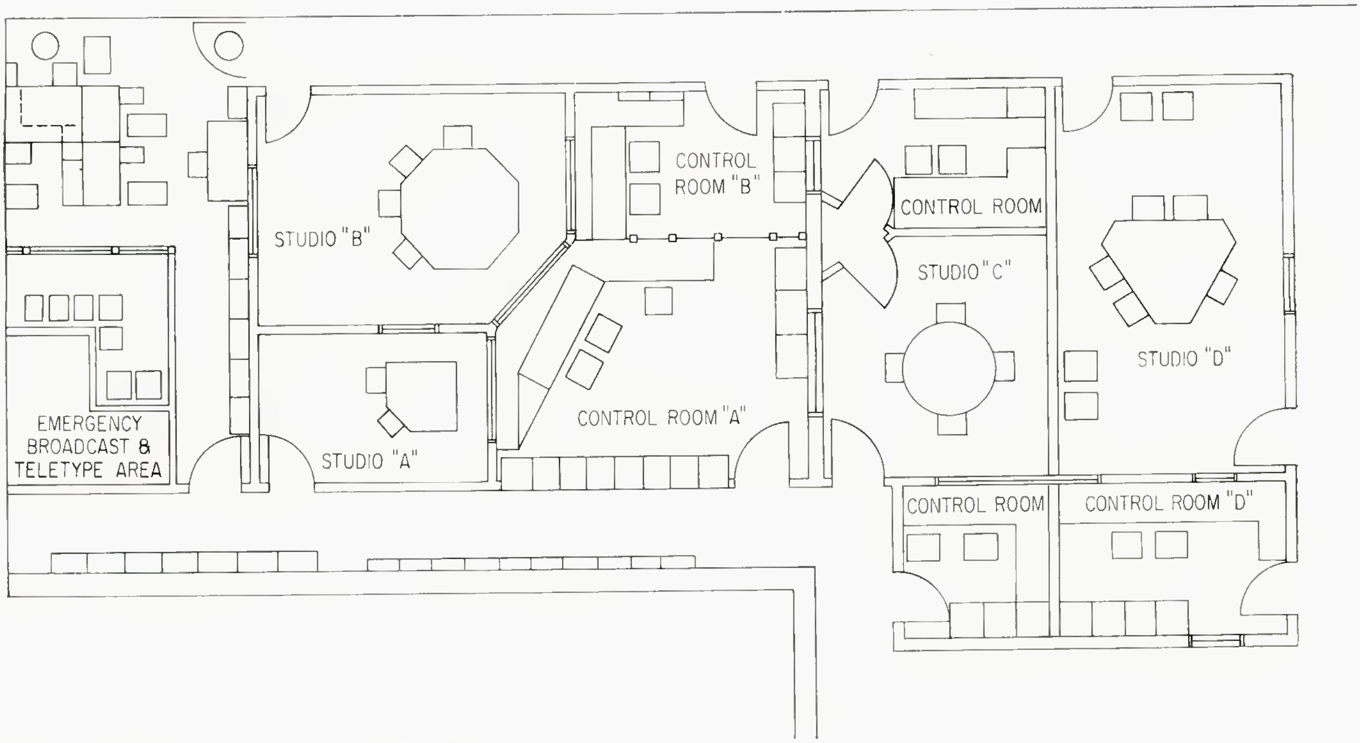

Figure 1. The floor plan layout of WCBS radio showing the studios and control rooms.

newscast. Code *two* is used similarly when the Washington news bureau is feeding the network line. If an exclusive news bulletin comes up, code *four* is sent and the program is fed following a ten-second delay to allow for breakaway. Code *nine* is used in the event of a national emergency, and in this case special alarms are actuated in the homes and offices of key station personnel. According to Mort Goldberg, director of technical operations of WCBS Radio, the Netalert system has proven to be virtually foolproof against accidental triggering in its more than ten years of operation.

To expand their coverage, in an area with a bewildering complex of news stories and events, WCBS operates three mobile units. These miniature studios on wheels are equipped with two-way radio, and mobile telephone communication. The base station of the two-way radio is on top of one of the world's tallest structures, the Chrysler Building. This is an important feature for reliable coverage because of the numerous tall buildings and structures in New York. Each mobile unit (there are two Ford Econoline Vans, and a Jeep Wagoneer) has a four-channel console and tape recorders with editing facilities. Normally a newsman and a technician man these vehicles. Reports can be broadcast live directly from the mobile unit, or a news spot can be completely prepared and the tape played back on cue.

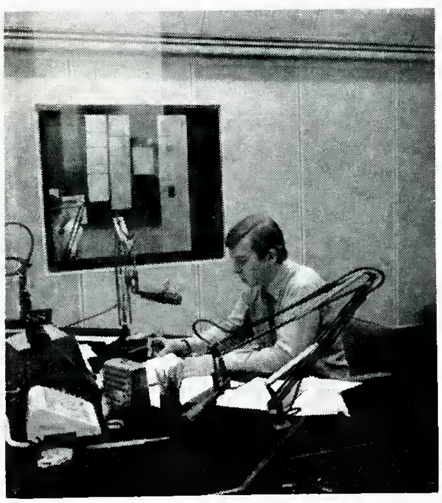

Figure 2. Studio B at WCBS. The control room is behind announcer Jim Harper.

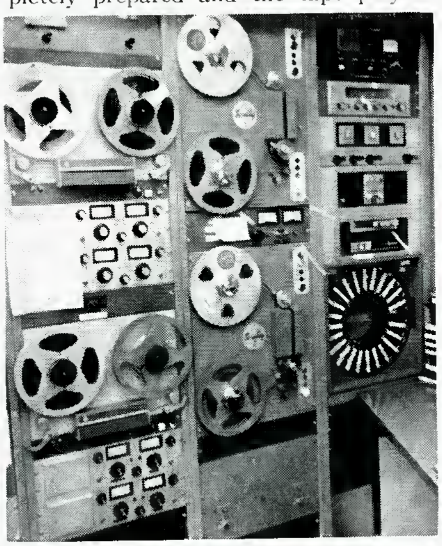

Figure 3. The racks in control room A. Metrotech logging recorders are on the left, f.m. automation Scully recorders are in the center, and a cartridge Carousel for the automatic insert of commercials and station i.d. are on the right.

The twice-daily rush-hour traffic syndrome is covered by a WCBS helicopter. In discussing the helicopter coverage with Mort Goldberg I mentioned the noise problem encountered by the police helicopter which reports traffic for the city-owned station. Here the cockpit noise tends to mask the pilot's reporting. I asked Mort how the WCBS pilot's voice was kept so far out of the noise. He explained that a special noise-cancelling microphone is used and that it is mounted virtually on the lips of the speaker. The pilot has a split headset, with one earphone monitoring the FAA channel for flight instructions, and one monitoring the WCBS two-way radio communication system. When the time approaches for the pilot's report, he switches over from the two-way radio to the output of an off-air receiver to get his cue.

The use of the helicopter need not be confined to reporting traffic conditions. On the recent parade of the Apollo 11 astronauts, the WCBS helicopter participated in a three-way coverage of the event. A mobile unit was dispatched to be in the parade, and was in communication over the two-way radio. Simultaneously the helicopter, with a two-man crew, hovered over the area and also was in communication over the two-way radio. A third form of coverage was provided by stationary positions along the line of the parade which were connected to the studios by telephone lines. The only thing missing from this array was a satellite in orbit—and we may have that before long.

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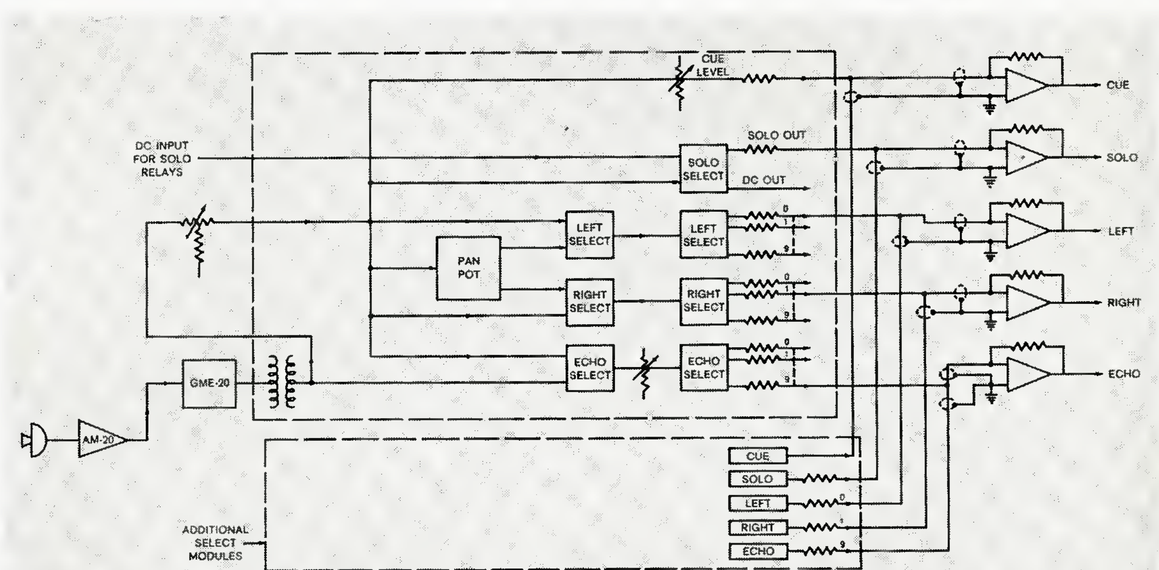

CSM Functional Diagram

Module provides capability of switching an audio input signal to 3 sets (left, right, echo) of 10 channel outputs for a total of 30 outputs ■ Internal pan control can continuously vary the signal level from left to right output ■ Echo and cue levels can be set independently and have separate outputs ■ Echo output can be switched for pre or post fader output ■ A momentary action push-button solo output switch provides for monitoring an individual module input ■ Input impedance: input isolation transformer provides an input impedance of 600Ω or 150Ω depending on primary connections ■ Insertion Loss: 1db MAX @ 1KHz when outputs are summed at an operational amplifier with a 6.2 K feedback resistor ■ Maximum input level: +20dbm ■ Cross Channel isolation is 80 db Min. ■ All level controls are audio taper ■ Pan control is 3 db down on both channels at center position ■ Power for lighting — 70MA at 24 to 30 volts D.C.; can also be ±15 V.D.C.

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Theory and Practice

NORMAN H. CROWHURST

● We must all live busy lives. Sometimes months go by without my receiving any technical queries of the type suited for discussion in this column, and I begin to wonder whether anybody has any queries and so fall back on bits and pieces I know haven't been covered lately.

Than last August's column took a different turn and I suppose several readers were interested to learn that I also write books. But a mere half dozen managed to find time to write and enquire further. Maybe more thought about it but then got busy. . . .

Next, in October's issue, the editor published a letter from a reader, asking how my book "Taking the Mysticism from Mathematics" could be obtained, to which the editor replied that I would autograph a copy for any reader who sent me $4.50 with his name and address.

Well, that brought a response, not only asking for that book, but whether I had any other books published, and also several technical queries that will be grist for this column. So rather than continuing where I left off last month, I'll take up some of these, because these are "live ones".

And in response to those who ask whether I have books on this or that, I've had a list of books still in print made up, which I'll be pleased to send free to anyone who sends me his name and address. Mine is P.O. Box 651, Gold Beach, Oregon 97444. The list gives a brief outline of each book, enough to show whether or not it includes anything of use to you.

Now here's a question that came with one of the book orders: Do I like emitter followers any better than I did the infamous cathode follower?

Well, that's a way of putting it! Many readers must have joined the scene long since I last joined issue about proper and improper use of cathode followers. And when I say joined issue, that brings up memories: a few people really did join issue with me, over a matter that is really a theory-and-practice question.

To answer that reader's question briefly, I'd say that, while there is a similarity between the emitter follower and the cathode follower, there are a couple of significant differences, that make it virtually a different animal. But get matters straight, let me briefly recap the cathode-follower situation.

When a triode tube, or some other tube strapped as a triode, is operated with the plate load in the cathode circuit and the plate connected directly to B+ (Figure 1), we have the "infamous cathode-follower" circuit.

This circuit has the property that, regardless of the d.c. plate resistor used, provided it achieves the operating mutual conductance of the tube, the source resistance, to an a.-c. signal, as seen at the cathode, is very approximately the transconductance of the tube.

Thus, if a tube with transconductance of 2,000 micromhos (2 milliamps per volt, to those who know that term better), is connected this way, the a.-c. resistance seen at the cathode is 500 ohms, which is the resistance equivalent of 2,000 micromhos.

Added to this, the textbooks pointed out that the voltage feedback in a cathode-follower stage is very nearly 100 per cent, so distortion is greatly reduced, as compared with the same tube operating with a plate-connected load of the same value. What the textbooks often omitted to point out is that this part of the theory assumed a normal plate load for that kind of tube is used, whether in cathode or plate.

My beef—which led to this notion that I "didn't like" cathode followers—was that you can't have your cake and eat it too. A suitable plate load for a tube with mutual conductance of 2,000 micromhos might be, say 20,000 ohms. This could give a voltage gain, with plate-connected load, approaching 40—say 35.

If, and only if, this is the a.-c. resistance connected in the cathode circuit, then a feedback factor of, say 36, is effective in reducing distortion. If the tube normally gives 5 per cent distortion at a certain level, working it as a cathode follower, with 20,000-ohm load, would reduce that to $5/36 = 0.14$ per cent distortion, at the same output level.

But—and here comes the beef—many people would use such a cathode follower to feed a 500-ohm load, because it provides a source impedance of 500 ohms.

While this is matching, of a sort, the low distortion often claimed doesn't follow through, as anyone who tried it found out. To start with, loading the same tube with a 500-ohm load probably raised the distortion to about 10%, without the feedback. And how much feedback is there?

With the figures used above, the plate a.-c. resistance of the tube would be about 2,850 ohms. Working this tube into an a.c. load of 500 ohms, makes the gain fractional, and the feedback factor about 1.17. So the 10 per cent is reduced only by this factor, to about 8.5 per cent which isn't so awfully good!

While I pointed out this fact, I also pointed out that the cathode follower had some very definite uses. If it was used in the output circuit of a preamp, connected to a high-impedance input, over whatever length line—but not long enough to cause mis-match problems—it could save hum problems just as well as any other form of 500-ohm line. But it must be terminated at the other end with a relatively high impedance (Figure 2).

Observing that little rule, the cathode follower could serve in audio circuits very well, and did. Just don't try and use it interchangeably with transformers!

But now, as I say, the emitter follower is something quite different again. The tube had an inherent value of mutual conductance (sometimes called transconductance nowadays), upon which its source impedance is based. Provided the tube was operated in conducting mode (not cut off or saturated) the performance would not deviate far from this.

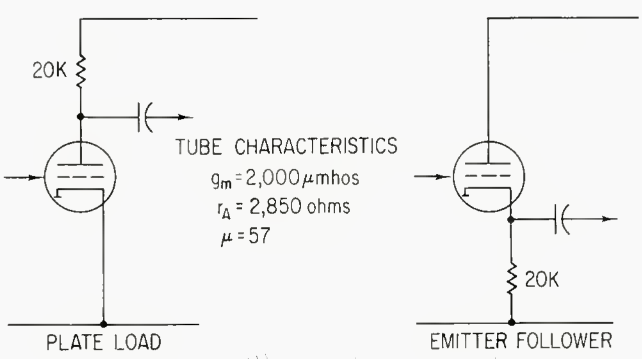

Figure 1. How the parameters of a tube produce the properties of a cathode follower (see the text).

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MODEL 1024. Commercial or personal stereo unit. Inputs Per Channel: Hi-Z microphone, mixing bridge, auxiliary bridge. Full mixing facilities. 1K ohm emitter follower output. Choice of speeds and head configurations.

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The emitter follower is much more flexible and has somewhat different properties. The important parameter about the transistors it uses is current gain, commonly known as beta. This means that collector current is approximately beta times base current, or conversely that base current is collector current divided by beta.

It also means (FIGURE 3) that the a.-c. at the base is approximately the a.c. delivered by the emitter, divided by beta, and that the a.c. delivered by the emitter is beta times the a.c. fed to the base.

As voltage transfer through an emitter follower is virtually zero, these changes in a.c. occur without changing the associated a.c. voltages.

Thus the emitter follower provides two-way impedance transformation. Suppose beta is 100, and that the a.-c. emitter load is 500-ohms: the a.-c. impedance reflected at the base will be 500 times 100, or 50K. And if the base circuit has an a.-c. source impedance of 50K, this will look like 500 ohms at the emitter.

This transformation is effective over a fairly wide range of values, and can be used in far more ways than was possible with the cathode follower. And feedback is effective, whatever values are used, calculated by methods we have outlined before, and probably will again.

So, to answer that question in a nutshell, I'd say that the emitter follower definitely does not suffer from the limitations to which I objected with the cathode follower. While I don't suppose I'll be designing any more circuits to use cathode followers, I have certainly used enough of them in the past. But nothing to be compared with the uses

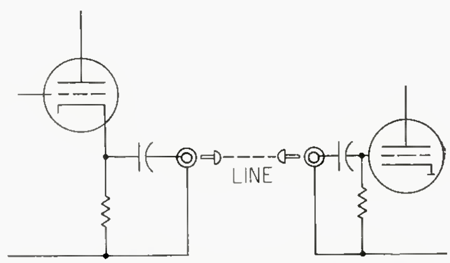


Figure 2. A valid way to use the old cathode follower.

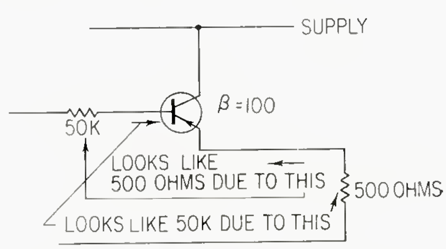


Figure 3. The basic properties of an emitter follower, which are far more versatile than the cathode follower ever was.

to which I have already put the emitter follower, with variations of which the cathode follower was not capable.

Incidentally, the responses to which I referred at the beginning of this column have led me to realize that the publishing scene today has a weakness: with all the junk mail coming out, promoting my books, and those of other authors who have something to say, buried in the pile somewhere, many readers cannot, or do not, find out about books they would really like to know about.

Also, authors have virtually no contact, or means of contact, with their readers. Many a reader would like to contact a book's author and ask a relevant question, but he doesn't know how. What I am now doing, in making a book-list available, not only enables readers to find out what I've written: they can write to me, too, if they have a question I might be able to answer for them.

And I might sell a few more books that way: I guess I deserve to. (*I think so too. Ed*)

ELECTRICAL SPECIFICATIONS

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

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

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

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

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

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

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

PHYSICAL SPECIFICATIONS

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

Cushions: Fluid filled for high ambient noise isolation.

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

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

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

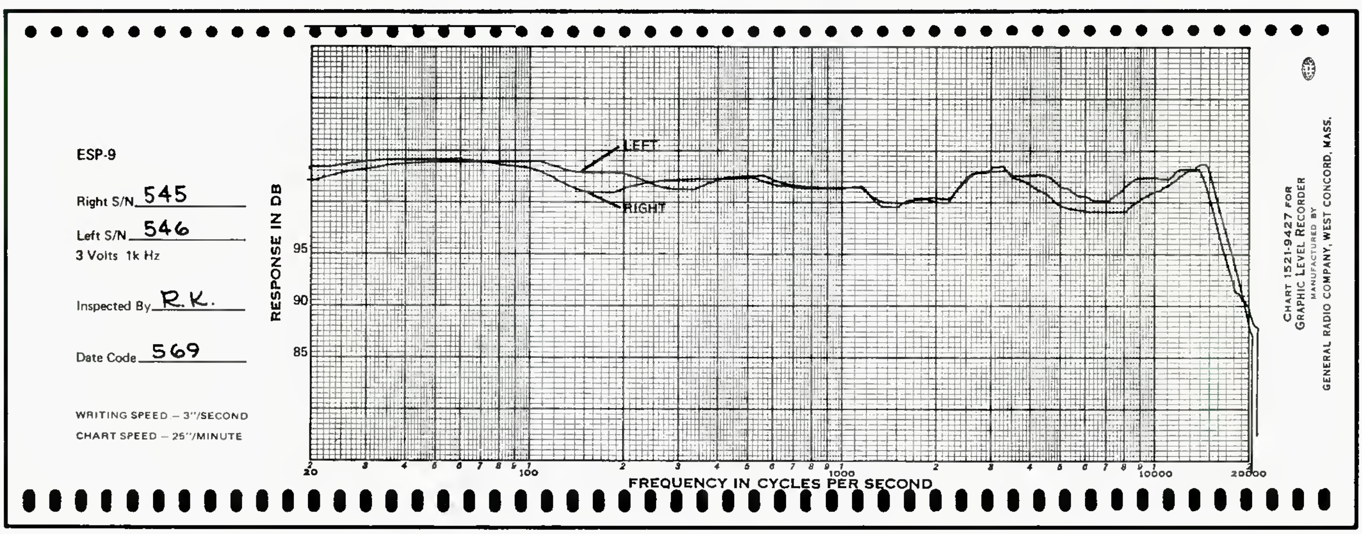
Headset Cable: Flexible, polyvinyl, 5 conductor, shielded, 6' long, black, with 5 prong plug keyed to E-9 Energizer receptacle.

Weight of Headset Only: 19 ounces

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

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

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



MACHINE RUN RESPONSE CURVE OF THE MODEL ESP-9

KOSS

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Primer on Methods and Scales of Noise Measurement

WAYNE RUDMOSE

Noise is with us all the time. We have discussed noise in the past and will continue to do so on the future. The understanding of basics to be gained in this article should be of considerable assistance to the reader.

Noise—in the sense of unwanted sound—has been a problem since Eve first poked Adam in the remaining ribs and told him to stop snoring. As time has gone on, and the population has increased, we have become even noisier—until today it seems that man-made sound of one sort or another bombards our ears almost continuously. The problem of noise has become so great that it is regarded as a special type of air pollution and, as such, is judged by many as a problem which might well lie within Federal jurisdiction.

In June 1968 a conference on noise as a public health hazard was held in Washington, D.C. It attempted to define the extent to which noise has become a public health hazard. Speakers from this country and Europe (where the problems are more extreme due to high population densities and where noise control is accepted as a matter of course) were called on for papers relevent to this problem. Many of these papers are of a highly technical character and not pertinent to readers of db. However, we feel that the paper that follows, by Wayne Rudmose, is outstanding as a basic summary of the methods used to measure the physical and psychological attributes of sound. It is, in fact, now being used at several universities to provide a more than basic level of understanding of sound measurements.

We are therefore pleased to offer this Primer—to appear in several installments. We wish to thank Mr. Rudmose and the American Speech and Hearing Association for their permission to reprint this material which originally appeared in the ASHA Reports 4 (February 1969). Ed.

It is appropriate to begin by commenting briefly on the nature of sound waves since many of the measurements under consideration are keyed to the fundamental properties of sound and because the reasons and necessities for using special procedures are intimately involved with the properties of the sound wave.

Sound, as we know it, is propagated in the form of a compressional wave. The fact that it is a compressional wave demands that the wave must move in a physical medium of some sort which can be compressed and rarefied. Sound waves are almost always referred to as compressional rather than rarefactional waves. We tend to overlook that, for airborne sounds, symmetry of waveforms ceases to exist when the amplitude of rarefaction reaches a peak value of one atmosphere, because there is nothing left to "rarefy." In a world where energy is sometimes measured in megatons of explosives, we must constantly remind ourselves that even for extremely loud sounds the total amount of energy in a sound wave is usually quite small. It follows then that the forces which sound waves can exert should be categorized as moderate to small. Only when one talks of extremely loud sounds will this classification be subject to modification.

Most of the sounds to be discussed here will be those propagated through our atmosphere. As the atmosphere is compressed and rarefied by these sound waves, a very small fluctuation in the barometric pressure, attributable to the passage of the sound wave, occurs. Although there are other measurable attributes of the sound wave, it is almost a certainty that, at this Conference, all of the data reported relating to the amplitude of the sound wave will have been measured by pressure-operated microphones sensitive to fluctuations in barometric pressure.

Two major classifications of sound wave types are used in acoustics: plane waves and spherical waves. These names are derived from the geometric description of the wave front, the area occupied by the forward part of the pressure disturbance. A plane wave results, in general, if the source is large, if the measuring point is at a long distance from the source, and if there are no reflecting surfaces of any kind between the source and the measuring point. Under these conditions the forward edge of the wave will lie on the surface of a flat sheet or plane which is perpendicular to the direction of propagation of the wave. Measurement of a plane wave's pressure is simplified because the microphone location is essentially unimportant as long as it remains in the plane or one parallel to it.

The spherical wave is classically described as the type of radiation coming from a small pulsating sphere where the point of measurement is reasonably close to the sphere and there are no reflecting surfaces in the vicinity of the source or the point of measurement. If one were to obtain an instantaneous snapshot of the pressure distribution of the forward part of such a wave, it would be evident that the maximum pressure, for example, would lie on the surface of a sphere with the center of the sphere located at the pulsating source. It should be pointed out that as one gets farther and farther from a spherical source, a reasonably small section of the forward wave front would occupy an area approaching a flat plane.

One important feature of a plane wave is that the pressure amplitude does not change with distance unless there is some form of absorption or dissipation in the medium as the wave travels through the medium. In the case of the spherical wave, the pressure amplitude decreases as the wave travels farther from its source, and this decrease is commonly referred to as the "inverse square law" change in amplitude. Since any real medium departs from an "ideal" medium, there is always some amount of dissipation of a sound wave as it advances through a medium. This dissipation represents a loss of energy, and the amplitude of the sound wave is affected by this loss of energy. Thus the amplitudes of plane waves do decrease with increasing distance and spherical waves lose energy at a greater rate than the inverse square law (6dB/ double distance) would predict.

FIELDS OF MEASUREMENT

It is important in measuring sound waves to recognize the various characteristics of plane and spherical waves. One must also recognize the interpretation of data depends not only upon the measurement of a sound wave's pressure amplitude but also upon the location of the measuring microphone. Points of measurement are commonly referred to as "far field" and "near field." These words are so descriptive that they need little interpretation except to explain one fundamental question: When is the measuring point in the far field and when is it in the near field? For simplicity let us assume that measurements are being made in an environment which is far removed from any reflecting surfaces. Such environments can be realized in a laboratory by providing highly efficient absorbing materials for wall surfaces, as in an anechoic chamber. In such an environment, a measuring point is considered to be in the far field if, by increasing somewhat the distance of the measuring point from a small source, the pressure measurement change follows the inverse square law, i.e., as the distance from the source is doubled the measured sound pressure level decreases by 6 dB. An engineering rule of thumb commonly used to specify a measuring position as being in the far field is that the measuring point should be at least several wavelengths, or several times the circumference of the source, whichever is larger, from the source.

The near field of a source is defined as that region where changes in pressure amplitudes do not obey the inverse square law. Normally such positions are within a few wavelengths of the source. A transitional region clearly exists between the near field and the far field as one increases the distance from the source to the measuring point. Unfortunately, the mathematical description of the wave in this transitional region is far from simple. In practice one tries to avoid making measurements in the transitional region, but in real life it is sometimes impossible to avoid making measurements within this ill-defined space.

RELATIONSHIP OF MEDIA TO SOUND PROPAGATION

At this point I wish to consider how the media affect the movement, or propagation, of sound waves, and to point out certain features of the waves that are altered when sound travels from one medium to another. Since a sound wave represents a pressure disturbance in a medium, and since this pressure disturbance moves through the medium, there must clearly be a certain speed of propagation of this pressure disturbance. The speed of sound within any medium depends upon the compressibility of the medium and upon the density of the medium. If the medium is solid, such as a metal bar, and the sound wave is being transmitted along the bar in the direction of its length, then the compressibility is defined by a term called "Young's modulus." If the medium is a fluid or a gas, the compressibility is given in terms of the bulk modulus of the fluid or gas. For most of the sounds to be discussed, the elasticity, or bulk modulus, of the medium is solely a property of the medium and does not depend upon the rate of compression or upon the amplitude of compression of the medium. Thus the speed of sound does not depend upon the frequency or amplitude of the sound wave but only upon the medium. The uninitiated cannot appreciate the simplification which results as a consequence of this statement, but if this were not true, then sounds would change in quality as a function of distance from the source, and waveform or shape would change as the wave moves etc.

As a sound wave travels from one medium to another, certain changes in the wave do, however, take place. It is important to understand these changes and to recognize how the characteristics of the media affect the changes. Since the frequency of a sound wave depends upon the source which created it; since the speed of propagation of a sound wave depends only upon the medium; and since the wavelength of a sound wave equals the speed of propagation divided by the frequency (a relationship which I just draw from thin air for lack of time), it follows that when the sound moves from one medium to another there is a change in wavelength. To illustrate the point, consider a sound wave traveling from air into water. Since the speed of sound is approximately four times as great in water as in air, the wavelength of a given frequency of sound in water will be four times as great as the wavelength of the same frequency in air.

Another important change that takes place as sound travels from one medium to another is a change in the pressure amplitude of the sound wave. As a sound wave strikes the boundary between two media, a reflected wave comes back with respect to the original wave and travels in the same medium as the original wave. In general, there will also be a wave transmitted into the second medium, and the pressure amplitude of this wave must obviously be less than the pressure amplitude of the original wave. Similarly, the pressure amplitude of the reflected wave must also be less than the pressure amplitude of the original wave, otherwise the law of conservation of energy is violated. The question of whether the pressure amplitude of the reflected wave is greater or less than the pressure amplitude of the transmitted wave depends in a somewhat complicated way upon the speeds of propagation of the waves and the densities of the two media. Crudely speaking, the more nearly the second medium resembles the first medium, the greater will be the pressure amplitude of the transmitted wave and the smaller will be the amplitude of the reflected wave; however, as the second medium becomes distinctly different from the first medium in the product of its density and the speed of sound, the reflected pressure amplitude will become greater than the transmitted pressure amplitude. This knowledge is important when one wishes either to absorb sound energy or to reflect sound energy ,and as subsequent speakers describe their subjects this fact will most likely become quite evident.

METHODS OF DEFINING A NOISE SOURCE

Turning now to the problem of how one defines a noise source, there are two commonly used methods. The first method describes the free field pressure radiation measured as a function of the angle of radiation from the source. Measurements using this method are normally made in the far field and represent the sound pressure radiated in various directions from the source under the condition that there are no reflecting surfaces in the vicinity of the source. Such measurements are commonly made in an anechoic chamber.

The second method leads to data which define the power radiated by the source. Measurements under this method are normally conducted either in a reverberation chamber or in a semireverberant room. Here the directional characteristics of the source are ignored, and the measurements yield only the total energy radiated. If one knows the free field radiation characteristics, the power of the source can be calculated; however, if only the power of the source is known, one cannot calculate the free field radiation characteristics.

The choice of method is made on the basis of the intended application. If one is interested in calculating the noise a machine will generate if it is placed in a room of known acoustical characteristics, then he should use the power definition. However, if one is calculating the effect of a machine as it radiates its noise toward a neighborhood, the free field pressure definition is generally most applicable.

In either case the measurements, whether they are sound pressure level or power level, are normally specified as a function of frequency. The most common way of specifying the frequency function is to supply data indicating sound pressure level in octave bands (or one-third octave bands) or sound power in octave bands (or one-third octave bands). It is conventional to use the decibel as the measurement of either sound pressure level or power level, but one must be careful to recognize that the reference levels for these two units are different. These reference levels will be discussed next month.

THE SOUND MEASUREMENT SCALE

Let us turn now to the physical measurement of sounds. Sound pressure is the attribute which relates to the amplitude of the sound, and frequency is the attribute which relates to the pitch of the sound. The range of sound pressures of interest to us is represented on the low end by the threshold of hearing of normal young people and on the upper end by the noise of small arms measured in the near field. Stated in physical terms, this sound pressure range is approximately from 0.0001 to 100,000 dynes per square centimeter (or microbars). (Atmospheric pressure represents 1,000,000 dynes per square centimeter.) It is clear that we are dealing with a tremendous range of sound pressures. Because in acoustics we are just as interested in observing the effects of small changes near the threshold of hearing as we are in observing the effects of small changes near the upper end of the scale, it would be impossible to construct a linear scale which would be applicable to our problem. An analogous problem which might be more meaningful would be one of measuring lengths where we are interested in having a scale ranging from one inch to 16,000 miles, and we need a ruler to measure changes of a few inches or changes of a few miles with the same ruler. The simplest mathematical scale available for either purpose is the logarithmic or decibel scale. One characteristic of the decibel scale is that it is possible to show, on an ordinary sheet of graph paper, a large range of sound pressures in such a manner that the small variations are as accurately portrayed as are the large variations. In my opinion this is the principal reason that the decibel scale is so useful in the field of acoustics.

Even though the decibel scale is a useful scale, it still creates many problems for the beginner and the uninitiated. I do not feel justified at this point in entering into mathematical definitions and calculations, but I shall try to give you some feeling for the units which you will hear and see in most, if not all, of the subsequent presentations. In our field of acoustics whenever the sound pressure level is designated in decibels, the reference level will be 0.0002 dyne per square centimeter (or microbar). On this scale zero dB sound pressure level corresponds to a pressure of 0.0002 microbar. A sound pressure level of 60 dB corresponds to a pressure not 60 times the reference pressure but 1000 times the reference pressure or 0.2 microbar. A sound pressure level of 100 dB corresponds to a pressure of 20 microbars, and finally a sound pressure level of 160 dB corresponds to a sound pressure of 20,000 microbars.

Sound-power level is also expressed in decibels, but here the reference level is 10^{-12} watt.[1] This means that a power level of zero dB represents a power of 10^{-12} watt. A power level of 60 dB represents a sound power of 10^{-6} watt; a power level of 100 dB corresponds to a power of 10^{-2} watt; and a sound-power level of 120 dB corresponds to a power of only 1 watt. These examples illustrate that for most of the sounds encountered in our daily lives the actual amount of power involved is less than 1 watt, which seems truly insignificant in comparison with our normal sources of power.

continued next month

A Different View of Speaker Coverage

ELLIOTT FULL

The choice of equipment to go into a public-address system should be made, at least in part, on the purpose for the system, rather than just on the basis for coverage. The author describes several examples.

The questions of psycho-acoustic considerations of a public-address system are brought up in this article. This is an area that has been explored extensively in print so far as we know. Accordingly, we would be most interested in hearing from other operators as to their experiences and practices to make a p.-a system do the really best it can.

"Keep the level low and use a lot of speakers"
"Put 'em in a circle and locate them high over the center of the crowd"
"Phase 'em and aim them at the crowd"

THERE ARE ALMOST AS MANY speaker placement theories as there are practitioners of the art. How a large crowd can be covered depends on many factors: What power will be needed to cover the background noise? What fidelity needed, will music be transmitted? Are there going to be reflections that will adversely effect understandability? Amplifier power, speaker efficiency and crowd density have to be calculated, sometimes empirically, and related to cover a crowd which itself may vary in size. How much money is available?

The above are some of the usual considerations that are, or should be, the concern of the p.-a. designer. Other factors are in the habit of cropping up. Each installation has its own problems.

Indoor p.-a. has many of the same problems, but the designer must be aware of the room or auditorium's time constant, so that the reverberation won't make the program material difficult or impossible to understand. Cardioid mikes and linear equipment will often help all p. a. systems, particularly the indoor variety.

There is one more problem that few, if any, designers account for. *What is the interrelationship between the audience and the man at the microphone*: is he just entertaining them, giving them their money's worth or is he going to demand something of them?

The premise of this article, admittedly only partially tested is that the audio for these two situations should be basically different. If the audio is to be used to warn of a hazard, such as the propellers at an air show, swimming pool precautions, race track dangers, unruly crowd warnings, etc., a different brand of audio should be used! In such cases we feel that the ordinary p.-a. system with its several medium-or low quality widely-separated speakers, restricted-range microphone and amplifier, will not elicit the audience response needed or desired. A commanding voice coming from one location, as with ordinary person-to-person communication, becomes more demanding, more real and less mechanical than other types of p.-a. coverage. This amplified voice becomes the "superman" that will take care of the problem situation. Nietze, the philosopher, outlined this concept in his "Man and Superman". If possible, this high quality, clear voice should be above or removed from the crowd. It should be used as little as possible.

Where unruly outdoor crowds are becoming more frequent, how would this theory be applied? First, a loud commanding voice cannot come from a squad car's fender mounted reflex trumpet. The *commanding voice* should be just as spoken by the man, but with the level raised about 30 dB or more.

Continuing with the police example, what hardware might be used? (1) A high-quality, medium-sized speaker such as a Klipsch La Scala mounted on a van, protected, water proofed and rotatable, (2) A good-quality solid-state 25-watt amplifier, (3) a broadcast-type cardioid microphone such as an EV RE15. The man at the microphone has to have a good voice and be trained and capable of using the tool to it's greatest advantage.

In a large auditorium with a high quality, low-level, multi-speaker installation, it might be well to shift to a central dominant speaker with amplifier gains and tone controls moved to marked locations in case of panic or fire. The announcer would then shift to an assured, controlled lower tone of voice, then point out the exits and minimize the concerns of the listeners. We have used this theory at several athletic contests. We bypassed the existing poorly-maintained multiple-speaker installation and installed an identical amplifier, well maintained, and a single speaker with a 26-inch mouth and four drivers. Among other requests made of the audience, we told them not to litter the grounds. After the all day events involving about a thousand people, the grounds were still clean when the spectators departed.

Elliott Full is vice-president of radio station KXIC-AM & FM in Iowa City, Iowa.

Decca's Vienna Venue

JOHN BORWICK

One of the worlds great serious music halls is also a recording studio (venue). The author guides us through this unique installation located in the city of classical music.

Surely the world's most famous recording venue is the Sophiensaal in Vienna. I have just been on a visit to Vienna and spent a fascinating time in the Sophiensaal at the invitation of Gordon Parry, Decca's senior recording engineer.

The Sophiensaal — there are three splendid halls in fact, superbly architectured *en suite* and decorated in Vienna's richest style — has hit the audio headlines in various ways. First, we know it is the place where Decca (London) recorded their monumentous complete cycle of Wagner's *Ring* — and numerous other recordings with the Vienna Philharmonic Orchestra, Willi Boskovsky, etc. Then a wider public saw Decca at work on recording *Gotterdammerung* in the historic film *The Golden Ring* made by BBC Television and since shown all over the world. And John Culshaw, formerly classical record director of Decca, has now set down in print the whole story of these Sophiensaal adventures in his book *Ring Resounding*.

Perhaps the most unique feature of this special "remote studio" is the thoroughness with which John Culshaw and Gordon Parry (as long ago as 1956, set about locating a suitable hall in Vienna and, having been delighted with the Sophiensaal acoustics) dug themselves in. They did this both technically and domestically, wiring the whole place for sound, designing and building a huge custom control console and even setting up house in an apartment of rooms upstairs. (This flat is the scene of many between-recordings parties and discussions and while lunching there I was very conscious of the great names who had preceded me, Flagstad, Solti, Nilsson, Karajan, Fischer-Dieskau, and so on. There were many momentoes too, like the actual steerhorns used in *Gotterdammerung*).

THE CONTROL CONSOLE

A general view of the massive control desk is shown in Figure 1. The now-familiar concept of slide fader modular strips each containing all the facilities for that channel was adopted. There are basically 20 Channels, in the unusual configuration of 10 to each operator position with the producer's chair (originally for John Culshaw) in the center.

This mode of working is a Decca speciality and it is common for Gordon Parry and his colleague Jimmy Lock to work in tandem, mixing the orchestra on one panel and voices on the other. Sometimes, as in the orchestral interludes in the recently completed *Der Rosenkavalier* recording, the whole desk will be used for the orchestra alone, even though the 8 switchable echo-return faders can be used as extra microphone channels. So a modification is planned, using space-saving integrated circuits, to increase the number of channels.

Four echo sources are on tap, two echo chambers and two stereo EMT echo plates, with remote-control reverberation period. The send/return circuitry is novel in that the return can be controlled either separately or ganged in with the channel fader. For truer orchestral reverberation (hardly ever needed) or special effects, the subsidiary Blauersaal can be co-opted as giant echo-room. It is effectively soundproofed from the main Grosse Saal and has an 8 seconds reverberation time. It is heard to impressive effect in the *Ring* recordings when Fafner's voice was bounced in there from 12 different loudspeakers! More subtly, I was given a foretaste of its use in the still-to-be-issued *Der Rosenkavalier* master tape where a beautiful airy perspective was achieved by placing the stage orchestra of some 40 players in the Blauersaal (with a separate conductor) and mixing this with the main performers in the Grosse Saal.

The distribution networks are unusually versatile. Outputs from individual channels, grouped in any required manner, can be sent to a variety of listening points. (It is important that the cue signal sent to off-stage performers does not contain their own sound.) Tape replay too is tricky, and can be stereo or mono and either selected or reduced from four-track as required. The Dolby noise reduction system is always used and this was certainly the first location in the world to install 8 Dolby units to give 4-track stretch and de-stretch (for monitoring) operation.

OTHER TECHNICAL FEATURES

Closed-circuit television is widely used. The control room has no window into the main hall and so the console faces two large t.v. monitors flanked by Tannoy loudspeakers (behind gauze curtains). British Quad 50E power amplifiers are used for line and speaker distribution etc., but the channel limiters and equalizers are all of Decca design.

With so many expensive and busy artists to record, great care has been taken to double up on equipment. The rack-

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John Borwick is technical editor of the British publication The Gramophone and is currently secretary of the Association of Professional Recording Studios with close on 150 member studios in Britain. He has contributed before on the professional scene in Europe.

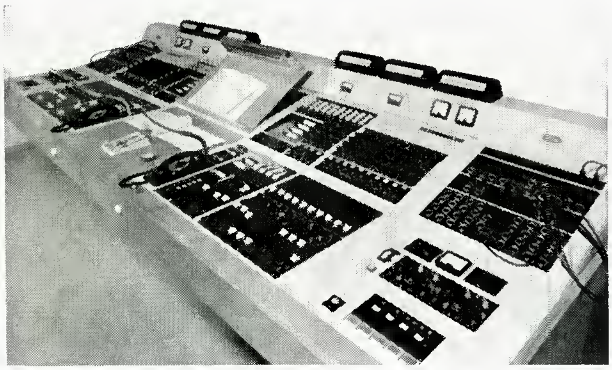

Figure 1. The console installed by Decca (London in the U.S.) at Vienna's Sophiensaal. Note the use of strip-type peak-reading meters.

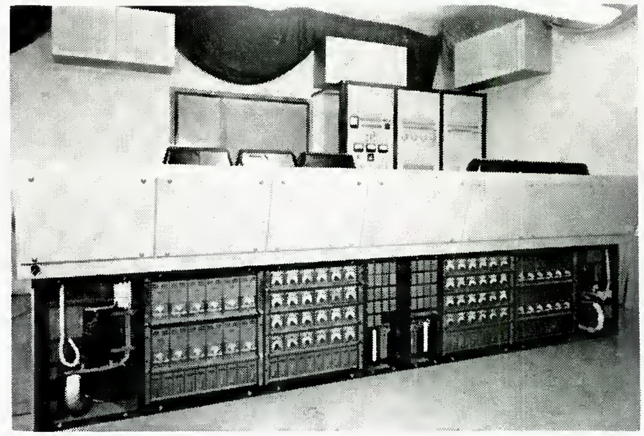

Figure 2. A rear view of the console shown in Figure 1.

mounted amplifiers, for instance (see FIGURE 2) have instant change-over switches and pull out for physical replacement. This is true of the power supply banks too, which identify faulty units by warning lights and can be replaced in a few seconds. Taping too is normally double-banked, using the EMT/Studer C37 and J37 studio machines so widely employed in European studios. Final editing of the tapes is carried out on the spot, so that it is the final *masters* which are sent back to Decca headquarters in London. They are Dolby-ized, of course, and so are compatible with the rest of Decca's classical record program.

Neumann electrostatic microphones have been first choice for a number of years but a revolutionary new microphone technique is presently being tried. This includes a trial of new AKG dynamic microphones and has been prompted by the need to look ahead to combined audio/visual presentations. As cameras and microphones will increasingly need to work together, re-thinking of the microphone's job is vital. Film and television needs have now been joined by EVR (electronic video recording) and Decca wants to be ready for any move towards the audio visual gramophone record of the future. During my visit, indeed, they were completing the sessions on a t.v. film soundtrack of *Cosi fan tutte*. The Decca engineers and the singers reluctantly agreed to voice dubbing for the arias but they felt that miming the recitatives was too prone to timing errors and so these were filmed and recorded simultaneously.

Nobody can measure the total effect on gramophone records (and equipment) sales brought about by the public impact of a major recording like the Decca *Ring* cycle. No doubt it is considerable, and clearly we can expect further treats and maybe surprises from Decca and from the Sophiensaal.

A Preamp/Balanced Modulator for Audio Experimentation

ROBERT C. EHLE

The author describes an audio frequency, balanced, amplitude modulator useful as a volume expander as well as an envelope shaper, gate, or bell generator and amplitude modulator for experimental audio and electronic music projects. A low-level phono preamp is included.

Many people are not merely content to turn their sound systems on and enjoy the music but must be constantly experimenting with various techniques in order to provide new musical experiences through the medium of high fidelity. To such people, high fidelity means more than faithful reproduction of a live performance and often becomes an attempt to apply electronics to a particular approach or interpretation of music. Involving themselves with synthetic stereo, enhanced reverberation, volume expansion, etc., they turn high fidelity into a creative medium and, if they continue far enough may find themselves actually composing original music for electronic instruments. This *electronic music* as it is called is a potential haven for all those musically inclined engineers and technically capable composers.

The balanced, amplitude modulator is a device which is capable of performing many special and interesting modifications to music and other sound material. It does this by a control signal automatically adjusting the amplitude of the program material to its own instantaneous changes in amplitude. If the control signal is a slowly changing, d.c. voltage the result is a slow change in amplitude of the controlled program. This type of operation is used in both the volume expander and the gate or envelope control mode of operation. On the other hand, if the control signal is an audio frequency range signal, the program material will be changed in amplitude at an audio frequency rate where each *loud* peak will correspond to the maximum amplitude of the waveform doing the controlling while the *soft* trough will correspond to the moment of minimum amplitude in the control signal waveform. This non-linear mixing or amplitude modulation results in the generation of a series of sum and difference frequencies from the interaction of the program and the control signals.

For many operations where amplitude modulation is to be used, a balanced modulator is desired so that the control signal does not appear in the output. With such an arrangement, the control signal may be constant but it will not be heard in the output while the sum and difference products will appear only when a program signal is fed into the main input.

The specific circuit designed by the author is shown in

Robert C. Ehle is a consultant and teacher of electronic music. He holds a master of music degree in composition and has taken advanced courses in mathematics, electronics, and computer-systems technology.

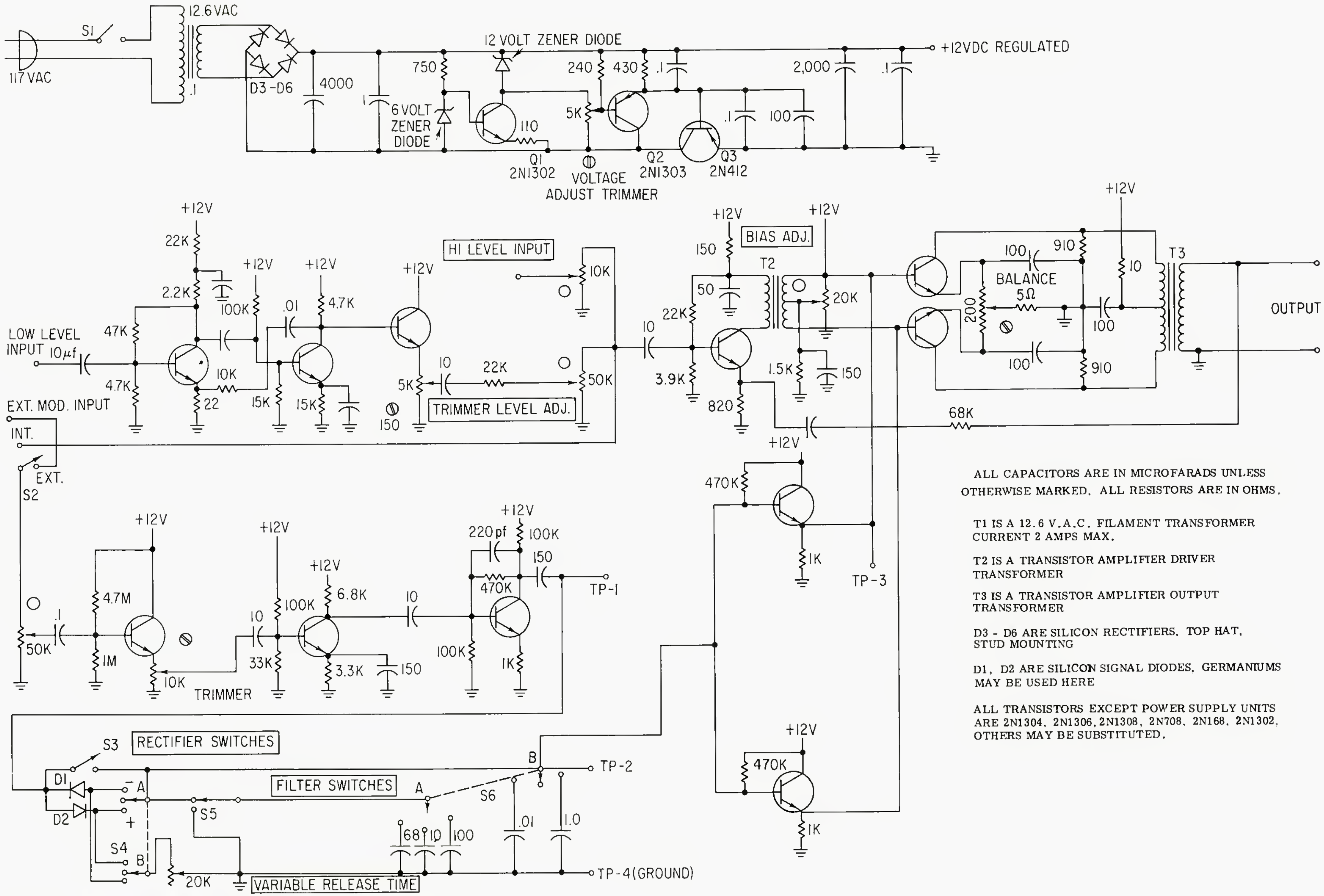

Figure 1. The solid-state balanced modulator circuit described in the text.

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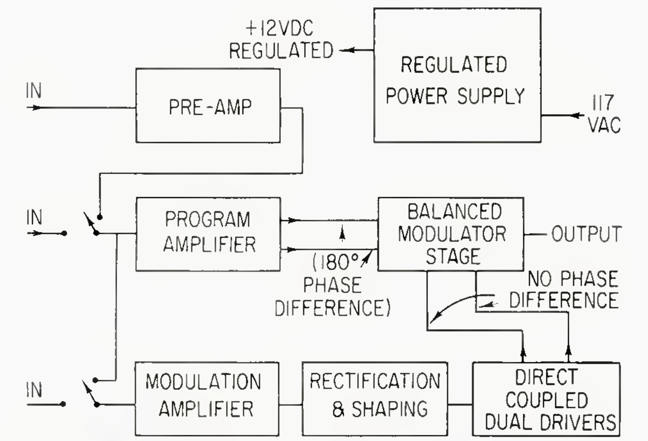

Figure 2. A block diagram of the complete balanced modulator.

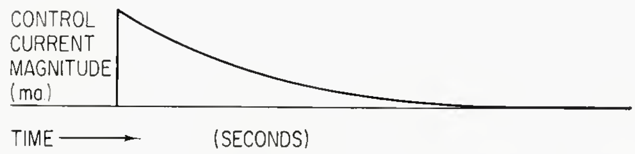

Figure 3. A graph of the current envelope used to generate synthetic bell-type tones.

Figure 4. The wave-form of pulse bursts, useful for testing audio equipment and other experimental applications including electronic music.

FIGURE 1 and 2, and is a solid state, amplitude modulator containing the following sub-sections:

1. A pre-amp for low-level signals, (this is a simple design; a more elaborate circuit with variable equalization might be substituted),
2. A modulation signal amplifier,
3. An envelope follower shaper with positive and negative rectification and six time-constant capacitors,
4. A balanced, amplitude modulator with one stage of program amplification and direct coupled, low impedance modulation drivers,
5. A regulated power supply.

Articles have been published previously concerning the design and applications of vacuum-tube designs for balanced, amplitude modulators.[1] In the design of a transistorized version, several new situations must be considered which require different approaches than were used with vacuum tubes. Perhaps the most important of these is the fact that since transistors are current operated rather than voltage operated as are vacuum tubes, we must provide a control current to the modulation stage instead of a control voltage which worked in the tube versions. In order to provide the necessary current, a low-impedance output driver stage must be used to drive the modulator stage. Since both halves of the modulator must be driven in parallel, yet a low impedance path may not exist between the two halves because they are 180 degrees out of phase with each other in the program material input, it is necessary to employ a pair of driver stages to drive the modulator.

In the volume-expansion mode, it is necessary to provide a relatively great direct current by means of a rectification and filtering of the audio input. In order to obtain a wide enough range in dynamic control, three stages of audio amplification have been used before the envelope shaping circuits. With this plan it is possible to drive the modulator hard enough that only the loudest passages are heard if such is desired. Such effects, while ludicrous in conventional music, may be useful to the electronic music composer who is working with unconventional material from the outset. The reader is referred elsewhere for information on electronic music equipment and circuits which may be employed with the modulator.[2]

On the schematic diagram of FIGURE 1, front-panel controls are marked with a circle while those best suited to be screwdriver adjusted are marked with that symbol — a circle cut by a diagonal line. The test points have been mounted on the panel and are used for metering points and, alternatively, as auxiliary inputs for special applications.

The most important and critical adjustment of the instrument is the balance control. This control is set by driving the external modulation input with an audio signal and adjusting the balance control for minimum output. The operation of the balanced circuitry can be improved if matched components, particularly transistors, are employed in the push-pull stage. Even greater improvement could be obtained by purchasing higher quality, balanced transformers. Cancellation actually takes place in the output transformer so this is a critical part of the balanced circuit.

After setting the balance control, the bias and the function switches are set to suit the various desired functions. For volume expansion, it was discovered that the saturation end of the output transistor curve was more linear and gradual than the cut-off portion of the curve which tended to be quite abrupt. The procedure for setting the unit for volume expansion is this:

Turn the bias control to a point where the output transistors are beginning to saturate. This point can be metered by a voltmeter at TP-3 and will be approximately 1.4 volts to ground; also the volume of a signal passing through the unit will be less than at some lower voltage measured at TP-3. Now set the switch, S-2, to internal and increase the modulation volume control to maximum. Open S-3 and turn S-4 to negative. Select a suitable time constant with S-5 and S-6 (fast for jazz and piano music, slower for orchestral music), increasing the shunt capacitance increases the reaction time of the circuit, hence, a longer time constant.

In the operation of the circuit in the expansion mode the amplified, rectified, and filtered signal is used to oppose the bias current of the output transistors, thereby bringing them out of saturation and into a range where their amplification is greater. Although a transistor is not particularly linear near saturation, this causes very little distortion if the signal amplitude at the input is kept small. Since negative control current causes an increase in volume by the mechanism described above, the diode switch, S-4, is turned to negative for volume expansion. Various amounts of volume expansion may be selected by adjusting the modulation volume control and the bias adjustment. The user should experiment with various settings of these and all the controls, particularly if he is interested in some of the unusual effects possible with this circuit.

Finally a discussion of some of the more unusual applications of the balanced modulator is in order. Most significant of these are the applications in electronic music and sound synthesis. One basic use of the balanced modulator is in the generation of complex tones. To do this, the external modulation input is driven by a set of continuously running oscillators. These signals will be balanced out in the modulator and will not be heard in the output as such; however, when a signal enters the signal input of the modulator, the output consists of the input plus sets of sum and difference frequencies made up of the input frequency and each of the modulating frequencies.

The unit may be used as a gate or envelope control device in many ways. If the signal which is to provide the gate is an audio signal, it can be driven into the external modulation input and rectified to produce the necessary varying direct current for increasing and decreasing the volume of the main signal channel. In this mode of operation, the set of the controls duplicates that for volume expansion with the exception of the modulation input switch set for external modulation. A more direct method of providing envelope control is to generate a varying direct current by some means such as a battery in series with a potentiometer (or a cadmium-sulfide cell) which is controlled by the operator, or by a mechanical device (such as a punched paper tape through which light passes to fall on the CdS light-sensitive cell). This direct current can be applied between test points *two* and *four* and the amplitude of the signal channel of the modulator will follow the variations in current.

Generation of synthesized bell tones (see FIGURE 3) is a simple matter with the device mentioned above. All that is required is a control current with a rapid rise and a gradual decay, obtainable with a push button in series with the battery and a potentiometer for decay control. Anyone interested in sound-effect generation or sound synthesis will find many more applications for the unit, for example, with a switch in series with a battery a tone-burst generator (FIGURE 4) can be devised. A tone-burst generator has many applications including the testing of the transient response of audio equipment. Driving test points *two* and *four* with a pulse generator gives a high-speed, electronic switch for audio signals. This arrangement is useful for testing transient response, recovery, and for synthesizing many special effects.

SUMMARY

This article has described a practical circuit for a transistorized, balanced modulator and, in particular, has stressed the adaptability and the versatility of the circuit. The unit described is capable of performing many functions including some remaining for the builder to discover.

Described is an audio frequency, balanced, amplitude modulator which is useful as a volume expander as well as an envelope shaper, gate or bell generator and amplitude modulator for various experimental audio techniques including electronic music procedures. A pre-amp is included for phono and other low-level signals.

REFERENCES

1. R. C. Ehle, A Complex Tone Generator for Electronic Music' **Audio Magazine,** October 1966
2. R. C. Ehle, Design of an Electronic Music Synthesizer, **db, The Sound Engineering Magazine,** November 1968

Sound with Images

MARTIN DICKSTEIN

● Just as in the audio field, where it is necessary to take advantage of (or circumvent) certain characteristics of the ear, it is also necessary in the visual field to know some of the operational shortcomings of the eye and then develop equipment and devices which perform in a manner which will provide pleasing sensations to the viewers.

The eye is complex enough to occupy a complete discussion by itself, but one characteristic which plays a most important part in the operation of various pieces of equipment is the eye's relatively slow reaction to light stimulation. Human vision can not distinguish two separate light impulses which have reached the eye in less than about 1/10th of a second. The principles of film projection and television are based on the fact that any succession of impulses reaching the eye in less than this time period result in the viewer seeing what seems to be a continuous and unbroken scene.

The simple stroboscope also operates on this principle. If a fluorescent light, pulsing at 120 impulses per second, were to shine on a rotating disc which had on it 2 pie-wedges painted on in a dark color for easy visibility, then it would be possible to tell how fast the disc was turning by the relative apparent rotation of the wedges. If the dark areas turned 180 degrees during a dark moment of the lamp, it would seem as though the disc were not turning at all. This would mean, then, that the disc was actually turning in sync with the pulsing lamp. If the pie-shapes seemed to be turning slowly in a direction opposite to the motion of the disc, the disc was turning slower than 60 revolutions per second. If the wedges were rotating forward, the disc's speed was faster than that of the lamp. The speed of the rotation of the wedges also indicates how far off the speed of the disc is relative to that of the lamp pulses. This results in automobile and stagecoach wheels on t.v. or film seem as if they are turning backwards while the car is moving forward. Of course, this principle is applied to testing the speed of turntables, tape recorder drive shafts, projector motors, many shop and factory rotary devices and is also used to observe fast repetitive motions at a slowed-down pace. Even the kids benefit from this effect when they look at the little books with animated figures on successive pages which, when flipped rapidly, seem to be showing a moving picture.

In the projection of motion pictures, a similar trick is played on the eye to provide what appears to be smooth movement in the images. However, the operation of the projector itself does not keep the film moving continuously past the light source port. The movement of the film is actually frame-by-frame past the light port with each frame stopping completely while it is being projected; the construction of the projector provides this motion.

A constant-speed rotary sprocket wheel pulls the film from the feed-reel in a steady motion. However, when the film gets to the film gate, it is intermittently jerked ahead by a claw which fits into the sprocket holes, pulls the film ahead one frame and then recedes, rises, projects, and pulls the next frame down, etc. It is, therefore, imperative that a loop be made in the film between the feed sprocket wheel and the gate to take up and release the quickly changing slack resulting from the two different types of motion and speeds feeding the film through the projector. Some projectors may have the claw mounted on a cam wheel imparting an eccentric drive to the claw shaft which is also pivoted at its center, while other projectors may have a Maltese-cross shaped device rotated eccentrically on a constantly rotating wheel—but the film motion has to be the same past the gate, frame-by-frame with a stop at each frame. To eliminate the flutter effect which would be visible if the film were to move past the gate with no precautions, a rotating shutter is inserted to cover the gate opening during the movement of film in the gate. A three-blade shutter is used most frequently, especially on the smaller film sizes, and the frame being projected is also obscured a small portion of the time it is in the gate to further prevent flutter in the image. Another loop is then formed by the film between the gate and the take-up sprocket wheel. The motion is, therefore, again smoothed out before the film passes over the sound drum and between the exciter lamp and the light-sensing unit. This provides low flutter sound from the optical track and also protects the film from being pulled too sharply by the take-up reel.

In the days of the silent movies, the film frames were close together and the spacing between them was very small. The aspect ratio of width-to-height was maintained and the frame spacing was kept to a minimum. When sound was introduced to the film, the width was decreased to make room for the track. This meant that the height had to be decreased in proportion to maintain the proper ratio.

It was discovered, however, that the aspect ratio considered proper did not really correspond with the true field of vision of the eye. Along with this, came the desire to improve the sound distribution. A new process was developed and called *CinemaScope*. When the film used for this process (35mm) was provided with the usual optical track, the picture width was optically squeezed on the film by a special lens but the height used all the available space to a very narrow dividing line between frames. The aspect ratio was, therefore, not the one that had always been used. It was filmed with an anamorphic lens which distorted the picture to a narrower width than usual. However, when the film was projected, the special lens on the projector compensated for this difference and the resulting picture was made wide-screen, more in keeping with the proper horizontal field of vision of the eye. To accomplish this expansion in the horizontal, the method used could be a lens system with the proper width-expansion factor built in, a mirror system which allowed the picture to expand by the normal reflective process, or with prisms which would spread the image depending on the angular relationship between them. To provide the viewer with distortionless images no matter where in the theater he sat, the screen is usually curved and this creates the illusion of having the viewer in the action. To further this illusion, the sound track can consist of four magnetic stripes on the film with three of them used for dialogue and music and the fourth (narrower than the others) for sound effects. The separate tracks can then be distributed to separate speakers located to enhance the involvement of the viewer. When four tracks are used, the film has two tracks between the image and the sprocket holes (one each side of the image) and two tracks on the outsides of the sprocket holes. In this case, the sprocket holes are made narrower to permit the image width to remain the same.

Another innovation in the film industry was the development of a process using 35mm film but covering a field of vision of approximately the same width as the normal human eye, about 146 degrees. This process, Cinerama, makes use of three projectors locked together in sync and shooting simultaneously onto a curved screen to provide a wide image. The two side projectors fire across the line of throw of the center projector to eliminate image distortion. In this process, the sound is recorded on six tracks and reproduced from screen and surround speakers in the theater. However, the sound tracks are on a separate blind film and played from a fourth projector which does not throw any visual image. Five of the tracks are voice and music with the last for effects.

A different method of producing wide-screen images makes use of the 70mm film, twice as wide as the other processes. Todd-AO uses one camera and one projector with a width of coverage of about 128 degrees. One advantage to using larger film is that there is less magnification required in projection with the result that the image is sharper. Sound is recorded on six magnetic tracks (one on each side of the image inside the sprocket holes, and two on each side of the film outside the sprocket holes). One interesting feature of this wide-screen projection is that the image can lose sharpness due to reflected light coming from the curvature of the screen itself. To prevent this, the screen is made with vertical ridges that eliminate spurious reflections and to keep the screen's reflected light directed to the audience.

As long as we are discussing various processes used to provide visual effects that are pleasant and exciting to the viewer, we should like to take this opportunity to add some previously omitted information. The omission was by no means deliberate. In our discussion of the color camera used on the recent Apollo flights to the moon, we mentioned that the color wheel utilized was an adaptation of the original color t.v. process developed by CBS. Actually, the process is still being used and further developed at CBS. The camera owes a good deal of its excellent functional capabilities to the latest developments of this process and the associated equipment and modifications made especially for the Apollo missions. We thank CBS for keeping us informed and up-to-date so that we may provide our readers with the latest and the best.

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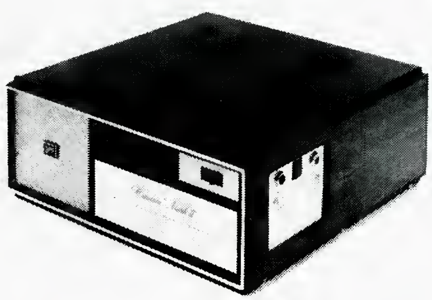

● Expanded programming and a new track-selection feature have been announced in this product, the Cantata Mark II, for the leased music field. A choice of on-location music systems is now offered to fit the budget and environmental desires of every type of prospect. The unit is offered for leasing with a basic package of several 1200-selection music libraries. The system also has a full line of features and accessories to permit operation with public address and payment of music royalties by the manufacturer. Operation is completely automatic, operation is designed for 24-hour use, automatic fading between cycles, and preset, non-tamperable controls.

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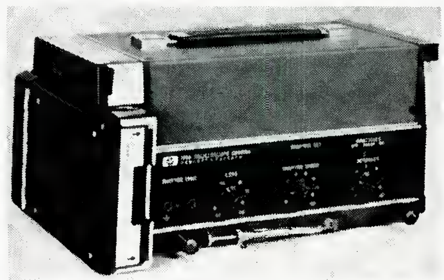

● There is no need to remove film and install focus plates when focusing this new camera. An external indicator eliminates the need for through-the-lens sighting. In addition the camera is adaptable to a variety of oscilloscope bezels so a single camera can easily accommodate several different applications. The new camera, model 198A uses a Polaroid back and standard flat-pack Polaroid films. Shutter and lens controls are readily accessible on the outside of the housing. Shutter speeds range from 1/60 second to 1 second. Shutter operation closes a contact so that the camera can trigger the 'scope or other equipment the instant the shutter opens. The shutter also has B and time exposure settings. The lens is a 75 mm 3.5 lens. A full 8 x 10 cm 'scope trace can be photographed.

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AMPEX AG-350-2 in MR-70 console, stereo half-track plus quarter-track play, 7½-15—$1850. Crown CX-744 4-track with sync panels—$1600. Ampex F-4450 (2)—$200 each. Set of 12 speed-change capstan sleeves for F-4450—$50. Pultec EQP program equalizer—$300. Fisher Series 80-C preamp/mixers (4)—$35 each. **R. S. Field, 833 Las Lomas, Pacific Palisades, California. (213)454-1050.**

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AMPEX 601-2 PORTABLE broadcast recorder, 200-ohm mike inputs, Samsonite luggage case, supply flutter filter, heads recently re-lapped for first time. With service manual and almost-complete spare-parts kit. Slightly marred case but in excellent operating condition: $350 plus shipping. **J. G. Holt, RD2, Box 314, Glen Mills, Pa. 19342.**

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ELECTRONIC FOR AMPEX 350 WITH POWER SUPPLIES FOR 1,2,3, or 4 track. **Box 11-B, db Magazine, 980 Old Country Road, Plainview, N. Y. 11803.**

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WHATEVER YOUR EQUIPMENT NEEDS — new or used — check us first. Trade your used equipment for new. Write for our complete listings. **Broadcast Equipment & Supply Co., Box 3141, Bristol, Tenn. 37620.**

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People, Places, Happenings

●The **Audio Engineering Society** has made its call for papers for the 38th convention to be held at the Los Angeles Hilton the 4th through the 7th of May. If you wish to present a paper at this convention check with the AES at its New York headquarters at 60 East 42nd St., New York, N. Y. 10017. Address titles and abstracts directly to the California chairman, **Hugh S. Allen, Jr., Gotham Audio Corp.,** 1710 N. La Brea Ave., Hollywood, Calif. 90046.

●**Bolt Beranek and Newman** are sponsoring a symposium on organ and church acoustics to be held on Tuesday, February 17th at the main Sanctuary of North Shore Congregation Israel at 1185 Sheridan Road in Glencoe, Illinois. The seminar will place particular emphasis on achieving fine acoustical environments for worship music while assuring high speech intelligibility. Organ design, room-acoustic design, sound systems, and noise control will be discussed. Among the speakers will be **Nils Schweizer** and **Harold Spitznagel,** architects; **Lawrence I. Phelps** and **Walter Holtkamp, Jr.,** organ builders; and **Robert B. Newman** and other members of the BBN staffs. There will also be a recital by **Margaret McElwain Kempber,** Dean of the North Shore Chapter of the American, Guild of Organists, on the temple's Casavant organ. Particulars may be had from BBN's Downers Grove, Ill. office at 1740 Ogden Avenue. The telephone is (312) 969-6150.

●**Advent Corporation** of Cambridge, Mass., has announced that it has reached an agreement in principle with **Dolby Laboratories** of London, England for the manufacture and marketing of a consumer version of the Dolby audio noise reduction system. The Advent unit will be a single-band device designed specifically to reduce background hiss in home-type tape recording. It uses the new B-Parameter circuitry. Advent expects to market its version in April of this year at a retail price of under $300. According to the announcement by **Stanley Pressman** Advent's vice-president for marketing, the product will make dramatic improvements in the noise level of and over-all dynamic range capability of any good home-type tape recorder.

●**C/M Laboratories,** manufacturers of audio amplification equipment has a new team of operating officers and directors, after an agreement was signed with an investment group to provide capital for the expansion and development of the company into a major factor in the professional quality field. New president is **Maury Jungman,** formerly president of **Belmont Electric Co., Inc.,** a division of **Arc Industries.**

Wayne Chou, the founder with **Nick Morris** of the company and recent president will remain as vice-president of engineering. Mr. Morris remains as vice-president production.

G. T. Thalberg has been appointed vice-president of sales and marketing. He was formerly a co-founder of **Benjamin Electronic Sound Corp.** He has also held executive positions with other high-fidelity manufacturers.

Among the new board of directors are such well-known figures in electronics as **Ira Kamen, A. B. "Burt" Covey,** and **John R. Poppele.** Mr. Jungman and Mr. Chou also serve on the board along with members of the investment group (not named in the release).

●**Hammond Hunt** is the new general manager of **Jensen Manufacturing Division of The Muter Company.** According to the announcement by **Herbert J. Rowe,** president of Muter, Mr. Hunt will be responsible for the total operations of the division. Prior to this appointment, he held several executive positions with Jensen.

●The **NAB** continues to grow. New membership figures for the new year include two more a.m. stations to a total of 2214, 30 new f.m. stations to a total of 1209 and 6 new t.v. stations to a total of 544.

●**Trevor Kendall** has been named president of the **Island Magnetic Electronics Corp.** The company is currently producing a line of production high-speed duplicating and peripheral equipment for the magnetic tape cartridge and cassette industry. A high-speed common-capstan tape duplicator is now being scheduled for early delivery along with supporting production tailoring equipment.

●A new company, **Satellite Film Service** has been formed to handle film strips, film editing, music scoring, video tape recording, tape duplication, and sync resolving. The owners are two brothers, **Warren and Fred Berney.** Fred has had extensive recording and film experience most recently with Academy-McLarty Productions of Buffalo, N. Y. Warren was formerly with the Army Missile Command dealing with the procurement of missiles for the Redstone Arsenal. He is a graduate of Temple University with a BS in Business Administration. The new company is located in San Jose, California and intends to service the San Francisco Bay area.

●**Michael Thaler** is the new sales manager for **Dubbings Electronics, Inc. Paul C. Smith,** president of Dubbings said that Mr. Thaler had been selected because of "his impressive background in the production and marketing of cassettes." Before joining Dubbings, he was with **Plastic-Ware,** a Bronx, N. Y. manufacturer of cassettes and parts. He was with this company for nine years. After that, and just before joining Dubbings he was vice-president of sales for **Allison Radio,** of Hauppauge, a manufacturer of stereo tape cassettes and cartridges.

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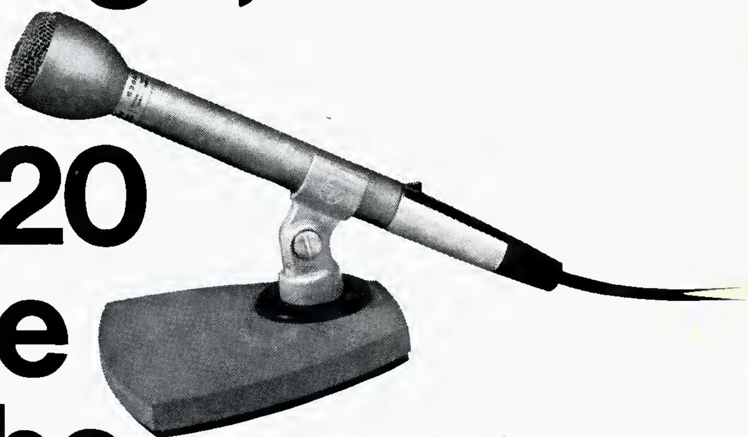

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A good little microphone, the E-V 635A. But just how good? After all, it was intended to replace the "workhorse" Model 635... a dynamic microphone that had earned its title under fire in studios and on remotes all around the world.

So when we introduced the 635A we put it to a critical test. A major recording studio was loaned a dozen 635A's and asked to test them. The engineers weren't told the price, but they got the idea that it was somewhere near $300.00.

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