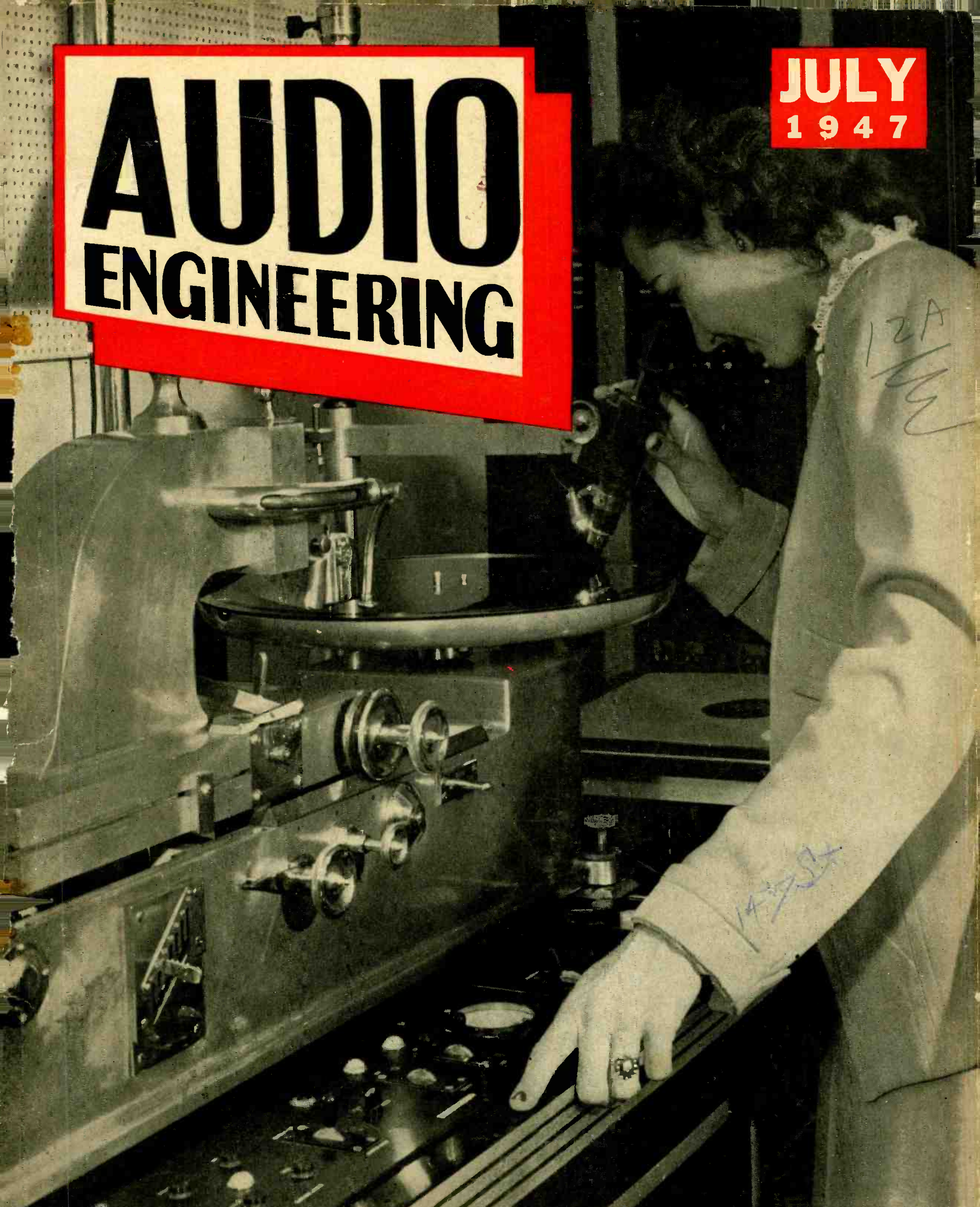
# AUDIO ENGINEERING

JULY 1947



THE JOURNAL FOR SOUND ENGINEERS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## CONTENTS JULY, 1947

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### COVER ILLUSTRATION

Operator at recording lathe. RCA cutter is shown.—*NBC photo*

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## Letters

### From a Famous Inventor

Sir:

You cannot imagine how delighted I am, first, because there is now a technical magazine devoted particularly to acoustics, but especially because your AUDIO ENGINEERING is so interesting and timely.

I am especially interested in your first two numbers because:

1. The Grieveson-Wiggins article mentions (*footnote c*) the pulse technique of acoustic measurement to avoid reflections, which I used for both microphone and sound source directivity measurements in 1919 (see my patent #1,507,081 on bilateral and unilateral microphones).

2. The Olson report on Listener Preference Tests, which reminds me of my own experience in 1921, while developing phono-electrical recording and reproducing apparatus for a large mid-west company.

After constructing a controllably-damped and soundproofed recording room, a 50-watt, high-quality amplifier, an electrical cutter head with over-cutting indicator, one-half inch diameter microphones with mixing control, and recorder circuits with equalizer and properly damped, resonant by-passes for the several formant resonances of microphone, cutter, *and* for the reproducer as well, I thought everything under good control and proceeded to make test records of such then prominent sound sources as Isham Jones' Orchestra, operatic soprano Claire Dux, pianist Godowsky, Chicago Opera radio broadcast pick-ups, etc. After many preliminary waxes I had the best of these converted to experimental pressings.

I had extended the frequency range from the usual 300-3,000 cycle range *with* resonant blasts at several points, to a range of about 100-5,000 cycles, with the resonant blasts pretty well removed.

So I asked the Directors of this company to come up to my laboratory for a look-hear demonstration.

I played a half-dozen or so of these assorted music type records, watching their faces grow first non-committal, then uncertain, and finally clouded with unmistakable signs of disapproval. But I had the temerity to ask: "Well, gentlemen, what do you think of them?"

The Chairman of the Board rose, wiped a little perspiration from his brow, and (probably estimating the cost of all these developments), with a sort of injured finality, said: "They don't *sound* like a phonograph!"

No, they most assuredly did not sound like a "phonograph", the only standard of musical sound these industrialists knew, but very much like the original musical sources which had been recorded, but with which they were not very familiar! This, as in Olson's tests, definitely shows psycho-acoustic conditioning by constantly heard inferior reproduction.

3. The Schlegel report on an FM Calibrator for Disc Recording Heads, which utilizes my FM Pickup System (see patents 2,273,975 and 2,319,622).

B. F. Miessner,
Miessner Inventions, Inc.,
R.F.D. 2,
Morristown, N. J.

# EDITOR'S REPORT

The story in *Time* magazine about S. Young White's article in our May issue evoked such a landslide of letters that *Time's* publisher ran the interesting letter reproduced on this page. We are happy to learn that this article had so much reader interest, and proud that it was picked up by *Time,* our favorite magazine. Our special thanks to Jonathan Norton Leonard, *Time's* brilliant Science editor, for his sparkling rewrite of White's story.


TIME

THE WEEKLY NEWSMAGAZINE

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INDEX



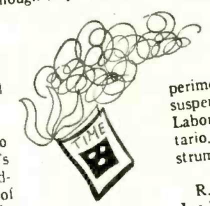



TIME, JULY 7, 1947


## A LETTER FROM THE PUBLISHER

*Dear Time-Reader*

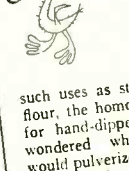
In the May 19 issue of TIME our Science Editor wrote a two-column story about a new ultrasonic gadget which could generate "silent" sounds powerful enough to set paper afire, audible sounds loud enough to paralyze strong men. It set up a chain reaction among our readers that is still snapping and crackling around these offices.



The story came to the Science Editor's attention via advance tear sheets of a new monthly technical magazine called *Audio Engineering.* Carefully checked with its author, Inventor S. (for Sidney) Young White, the story described the gadget, told what it could do, and suggested some aspects of its possible ultrasonic future —such as killing bacteria, breaking up suspensions of solid particles, precipitating smoke and dust, etc.

The morning after TIME hit the newsstands the Science Department's telephone got busy with readers wanting to know more about the ultrasonic siren and its inventor. The Public Library called to say that they were getting inquiries for *Audio Engineering* but couldn't locate it on any of their lists. The president of Ultrasonic Corp. turned up in person to get more information direct from us. He had read the story on the night train from Boston and had stopped by our office on arrival.

In the next few weeks this initial reaction grew to startling proportions for Inventor White, *Audio Engineering* and TIME. The radio and sound recording fields were heard from almost *en masse.* Atomic scientists conveyed their interest, as did numerous industrialists, colleges and universities, foreign scientists, etc. Some wanted to know how to manufacture the ultrasonic siren; others asked whether it could be devoted to such uses as sterilizing insect eggs in flour, the homogenization of chocolate for hand-dipped candies. An invalid wondered whether the instrument would pulverize his kidney stones without damaging him. The Long Island Duck Farmers Association thought it would be ideal for de-feathering ducks. Some others who have been heard from to date:



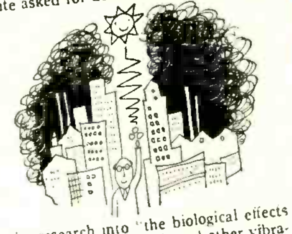
Dr Lee de Forest, of American Television Inc., wrote in to ask whether Inventor White had applied his siren to remedying afflictions like tinnitis (ringing in the ears), which has plagued the "father of radio" for years.

Consolidated Paper Corp., of Montreal, Canada, asked if the siren could be used to combat bacteria in paper stock and water.

Aluminum Ore Co., of Mobile, Ala., invited White to experiment at its plant in breaking down suspensions of solids, etc.; Aluminum Laboratories, Ltd., of Kingston, Ontario, desired complete data on the instrument's application to metallurgy.

R. R. Donnelley & Sons Co., TIME Inc.'s Chicago printers, wanted to know if it would be possible to "set ink on paper by molecular vibration through a sound medium."

The Naval Medical Research Institute asked for detailed information for its research into "the biological effects of intense sound fields and other vibrations."



Brewers, dentists, laundries and jewel graders have also made inquiries. A hospital hopes that the siren can be made to sterilize the air in its infants' nursery. General Mills would like to know whether it will coagulate the dust that plagues the packaging of its dry cereals. Cincinnati is ready for an immediate demonstration of the power of ultrasonics to lay the smoke and dust for which that city is renowned.

All in all, our Science Editor is very happy over this fresh evidence of the readership his department has, and Inventor White, one of the few independent U.S. research scientists, is convinced by the quality of the inquiries he has received that he has made an original, useful contribution to science.

*Cordially,*

*James A. Linen*

9

In our next issue, Winston Wells will start a series of articles on electronic musical instruments. For a long time, there has been a need for such a series, and we feel fortunate in obtaining such a thorough coverage of the subject by one who has specialized in electronic organ design for many years.

—*J. H. P.*

# Planning a Studio Installation

J. D. COLVIN*

**PART I—This is the first of a series covering broadcast studio installations. The methods outlined are also suited to large public-address projects.**

When a newspaper publishes a story covering the opening of a new broadcast studio plant, the article almost invariably mentions that so many miles of wire were used in connecting the equipment together, requiring many thousands of soldered connections. Such startling figures impress the public with the complexity of a broadcast station, but not in the same degree as did these same miles of wire and thousands of connections impress some engineer, causing him to spend many weeks and sometimes months of long hours in determining where it all went and what piece of wire was to be soldered to where. By the time the last wire is pulled and soldered into place, he is the one who is most thoroughly convinced of the complexity of the installation. He is the one who is most jubilant when the equipment is finally turned on and everything tests O. K. He is the one who is somewhat crestfallen when test shows that the twenty-one pairs connected to the monitor terminal block have been attached in reverse order and must be changed. He is the one whose face grows red when the brand-new hi-fi speaker for master control is unpacked and he discovers there is no outlet provided for its connection, and that the omission of this detail means chopping a trench across ten feet of concrete control room floor to run a conduit to the speaker location.

Radio engineers are human beings (despite the contrary opinion of some architects) and as such may make errors. However, the errors made run in an inverse ratio to the amount of thought and planning spent on any one job, and with the amount of experience in the type of job being handled. No matter how large the job may be, breaking the planning of the job down into its proper sequence eliminates its seeming complexity and reduces the possibility of errors in the final construction and installation plans and also reduces the time involved in the preparation of these plans.

### First Steps

It is the intent of this article to outline a sequence of planning a studio installation that has been found to proceed in the most logical manner and that automatically takes into account all circuits originally planned and their conduits. Each step in the planning will then be treated separately, the steps combined and the plans completed. It should be pointed out that the article covers only the audio facilities of the studio layout, and is not in any way concerned with the architecture of the building other than what may be involved in the placement of equipment and the running of conduits. Since the

*Audio Facilities Engineer, American Broadcasting Co.

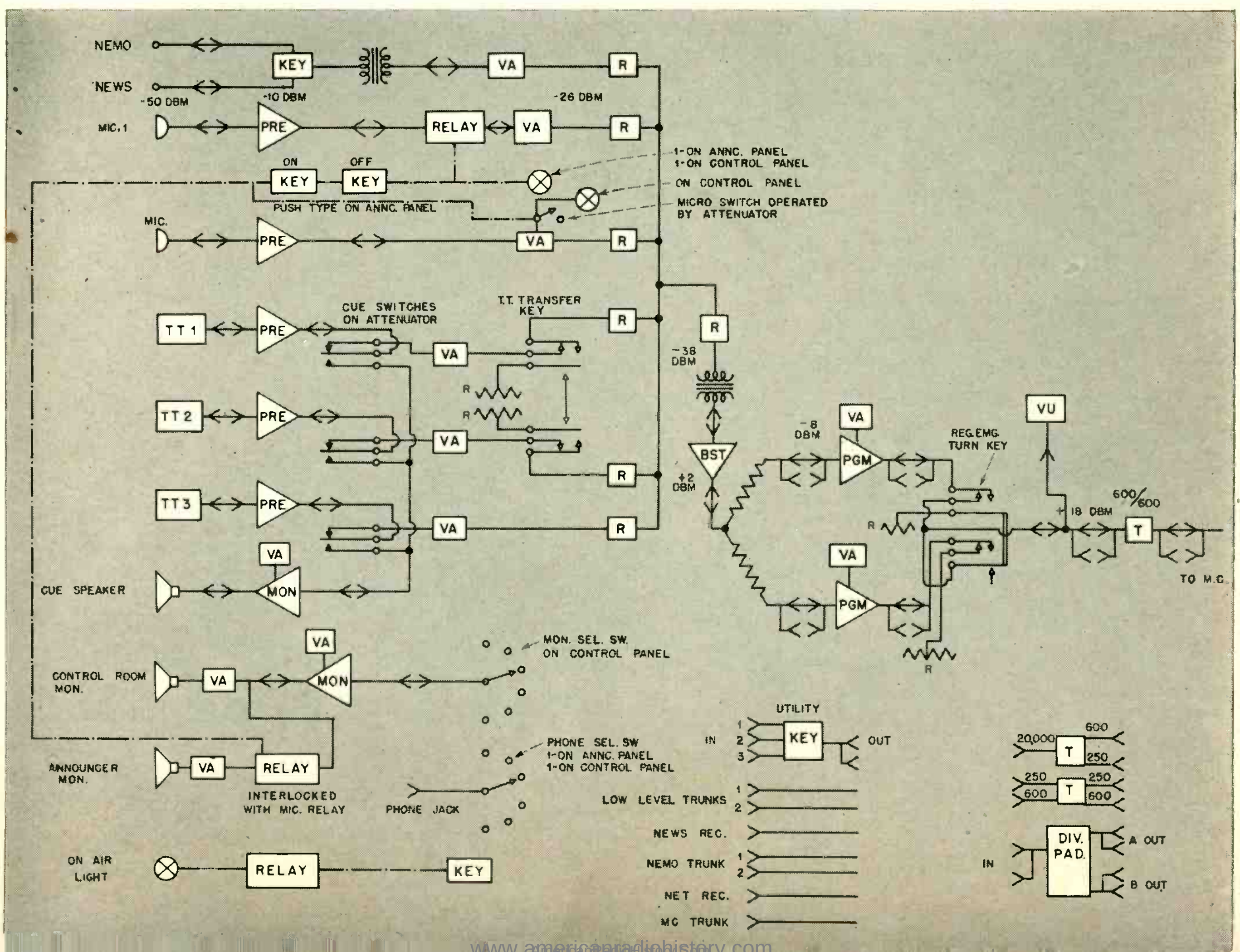



Fig. 1. Block diagram of an actual studio layout.

| | | | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|---|
| LL TRK 1 | MIC 1 OUT | MIC 2 OUT | T.T 1 OUT | T.T 2 OUT | T.T 3 OUT | MIXER OUT | DIV PAD REG OUT | MULT PAD REG OUT | DIV PAD EMG OUT | MULT PAD EMG OUT | MIC 1-2 RLY OUT |
| LL TRK 2 | PRE 1 IN | PRE 2 IN | PRE 3 IN | PRE 4 IN | PRE 5 IN | BOOSTER IN | PGM AMP REG IN | MULT AMP REG IN | PGM AMP EMG IN | MULT AMP EMG IN | MIC 1 MIX IN |
| NEMO NEWS KEY OUT | PRE 1 OUT | PRE 2 OUT | PRE 3 OUT | PRE 4 OUT | PRE 5 OUT | BOOSTER OUT | PGM AMP REG OUT | MULT AMP REG OUT | PGM AMP EMG OUT | MULT AMP EMG OUT | SPARE |
| NEMO NEWS MIX 1 IN | MIC 1 RLY IN | MIC 2 MIX 2 IN | Q-SW MIX 3 IN | Q-SW MIX 4 IN | Q-SW MIX 5 IN | DIV PAD IN | KEY REG IN | MULT KEY REG IN | KEY EMG IN | MULT KEY EMG IN | SPARE |
| REG EMG KEY OUT | MULT KEY OUT | SPR DIV PAD IN | MULT PAD IN | NEWS RECEIVE | NEMO TRK 1 | NEMO TRK 2 | NET. RECEIVE | MC TRK 1 | SPARE | TT Q-SW OUT | MON SEL OUT |
| LINE COIL IN | MULT COIL IN | SPR DIV PAD A OUT | MULT PAD A OUT | KEY NEWS IN | KEY NEMO IN | STD 1 TRK 1 | STD 1 TRK 2 | VI IN | SPARE | Q AMP IN | MON AMP IN |
| LINE COIL OUT | MULT COIL OUT | SPR DIV PAD B OUT | MULT PAD B OUT | SPARE | SPARE | SPARE | SPARE | MULT | MULT | Q AMP OUT | MON AMP OUT |
| MC STD 1 IN | MULT MC IN | SPARE | SPARE | SPARE | SPARE | SPARE | SPARE | MULT | MULT | Q SPKR IN | MON SPKR IN |
| T1 BRDG IN | T1 250 OUT | T1 600 OUT | T2 250 IN | T2 600 IN | T2 250 OUT | T2 600 OUT | UTIL KEY 1 IN | UTIL KEY 2 IN | UTIL KEY 3 IN | UTIL KEY OUT | KEY OUT MULT |

Fig. 2. By using a simple chart as is illustrated above, the layout of the jack field is greatly simplified.

method of planning a studio installation is the same whether one or a number of studios, including a master control are involved, a single studio was chosen to illustrate the method for the sake of simplicity of the drawings required.

This article takes up at the point where the management of the station, which includes the chief engineer, has decided upon the number, size and purpose of studios to be included in the plant, and all associated equipment such as turntables, recorders, offices to be supplied with monitoring facilities, etc. and the project turned over to the chief engineer for execution into the finished product.

The steps in planning the installation are:

1. Preparation of the single line block schematic covering all the facilities.
2. Preparation of Jack Fields, Amplifier Racks, and Console Panel and Cabinet layouts.
3. Preparation of a preliminary conduit layout.
4. Preparation of material lists of those items such as not shown in items 2 or 3.
5. Preparation of "running" or connection sheets for Rack and Console wiring.
6. Preparation of interconnection sheets for cross connecting of all equipment assemblies together.
7. Preparation of final conduit layout.
8. Follow up on actual work of construction.

### Preparing the Single-line Schematic

The block diagram is a simple method of putting the audio facilities in their entirety on paper. It provides a means to visualize the circuits involved, and on which such components as jacks, switches, pads, attenuators, amplifiers, line coils, etc. can be properly placed. The input and output impedances of the various components can be decided upon. The circuit gains and circuit losses can be calculated and indicated on the diagram. The circuit itself can be changed about until the best arrangement is obtained to meet the operational requirements. It is the proper place to make and correct all the mistakes that are to be made on the job, since an eraser is the only tool required to make the change.

*Figure 1* shows a block diagram of an actual studio layout that was designed to handle a specific program requirement. It will serve as a good example for the present discussion since it involves practically all of the elements that go to make up studio installations—microphones, turntables, speaker interlock relays, push-button control of the master switching relay system and spare facilities. This diagram as shown in *Fig. 1* is now in its final form, but did not arrive at this state without a number of changes in such things as jack locations, arrangement for switching in the emergency amplifier, pad values and the addition of several transformers. The studio involved is intended to handle the playing of delayed broadcasts, local transcription spots, announcement of the station call letters, occasional news broadcasts and the handling of a nemo circuit or a feed direct from the news room. Thus, there are the six inputs to the mixer system, Nemo—News, Mic 1, Mic 2, TT 1, TT 2, and TT 3; all feed into preamplifiers except the Nemo-News input which is of sufficient level. The output of the two microphone preamplifiers feeds directly into their respective attenuators through an interlock relay. The output of the three turntable preamplifiers feeds into the new Daven-type attenuators having the Cue-Switch feature. Turning these attenuators to the "off" position operates the switches designated as "Cue Switches" and connects the output of that turntable directly into a Cue amplifier and allows spotting of the transcription on that turntable. Since TT 1 and TT 2 are the two turntables intended for the delayed broadcast operation, in which it is often necessary to make an instantaneous switch from one machine to the other, the lever key following their attenuators was provided. The key thrown to one side connects the associated turntable through to the mixing network and at the same time disconnects the other machine. Throwing the key to the opposite position reverses the arrangement. Thus when TT 1 finishes, a quick throw of this switch to the opposite side disconnects TT 1 and connects TT 2 into the mixer circuit. This same switch in its neutral position connects both turntables through and permits segway with the attenuators as is sometimes necessary.

Since a one-sided grounded mixing system was employed, and since it is desired that the input transformer on the booster amplifier (which is of the same type as the preamplifiers) remain balanced for patching reasons, it became necessary to insert an isolation transformer between the mixing system and the booster amplifier. Advantage was taken of this necessity by making the transformer match the output impedance of the mixing network to the input impedance of the booster amplifier with the consequent saving of several db of matching loss.

Leaving the booster amplifier, the circuit divides through a pad and feeds into a regular and emergency line amplifier. The output of either of these two amplifiers is selected by means of the regular-emergency key that is located on the control console. The object of this arrangement was to make possible a quick switch to the emergency line amplifier in case the regular failed without the necessity of patching. This arrangement is somewhat superfluous since it is equally as possible that the preamplifier being used by the turntable on the air might fail thus making it necessary to patch in a spare preamplifier. From the output of this regular emergency key, the circuit feeds through a line coil to Master Control.

It will be noted that normally-through jacks are provided on the input and output of all amplifiers to permit patching in case of amplifier failure or to provide for other than normal circuit arrangement. Multiples are provided on those jacks having sufficient level to operate headphones as an aid in trouble shooting. There are cases in which multiples are provided on lower level jacks, such as preamplifier outputs as a means of picking up a feed for studio sound reenforcing systems, sound effects, etc.

## Relay Interlock Circuits

One thing which a single line block schematic cannot show clearly is the detailed operation of relay interlock circuits. The best it can do is to serve as a reminder of what exact operation the engineer had in mind when he comes to work out the two-wire diagram of the circuit. In the diagram under discussion the dash-dot line indicates the control circuit for the relay operation. Briefly, the operation of the arrangement is to turn off the announcer's monitor speaker whenever Mic 1 or Mic 2 are alive. Mic 1 is turned on or off by the announcer by means of ON-OFF push keys. (The control engineer rides gain at the control console.) Mic 2, used less frequently, is turned on simply by opening the Mic 2 fader. A microswitch mounted on the cover of this attenuator closes just as the attenuator is turned from its "off" position. Closing of this microswitch operates the relay system and kills the announcer's speaker. Indicator lights on both the announcer's and control engineer's panels indicate when Mic 1 is on, and one light on the control panel shows when Mic 2 fader is open.

Program Cue is obtained by means of rotary switches that pick up the monitor circuits from Master Control. Separate switches are provided for loudspeaker and for headphone operation. This is especially important for the announcer, who can listen to cue over his headphone when his mike is hot and speaker is off.

Such odds and ends as the required trunks to master control, spare coils, pads, utility key, etc. are all shown on the diagram. The "on air" key, "on air" light relay and "on air" light are also shown. Thus everything that goes into this studio setup is shown without particular thought at this time as to where each particular part is to be located—in the equipment rack or control console. This will be decided when their units are laid out.

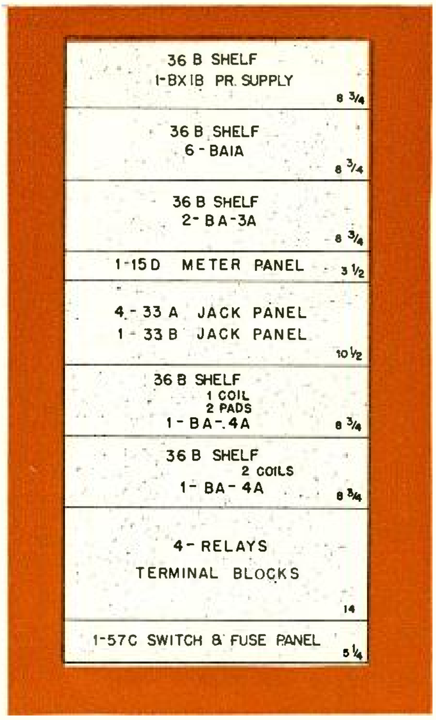



**Fig. 3. Rack-and-panel layout.**

This rather long explanation of *Figure 1* was carried out to show what reasoning and thought must be applied to its preparation. Actually the process here is in reverse—the block schematic was explained to show how the circuit functions are drawn, although the reasoning and thought come first and the block schematic is the result.

## Equipment Layout — Jack Field and Rack

Having all the circuit functions outlined on the block diagram with all the necessary components accounted for, the next step is to prepare the physical arrangement of the equipment. Functionally, the separation of those components that go into the equipment rack and those which go into the control console is fairly simple. Only those units that are used to control a program on the air, such as attenuators, lever keys, cue selector switch, monitor volume control, etc. should go on the console. All other units such as amplifiers, jacks, line coils, etc., go into the equipment rack.

Beginning with the rack, it is essential that the jack field be prepared first in order to determine the exact number of jack rows and the space required for them. An aid to laying out the jack field is shown in *Fig. 2*. It consists simply of a ruled chart with a block to represent each pair of jacks—twelve per row as is the case of the standard jack field. Since double jack rows were intended to be used on this job, the rows of blocks are paired to simulate the actual jack strip. These blocks are now filled in (in pencil with an eraser handy) with suitable abbreviated designations until every jack shown on the block diagram is taken into account.

As to the arrangement of the jacks in the jack field the following simple rules can be followed:

1. Group together circuits that do not differ in level by more than 30 db.
2. Attempt as far as possible to have circuit "outs"—such as Mic "outs", TT "outs" all in one row. Immediately beneath each such "outs" in the next jack row the following circuit inputs are located—such as preamp "in" etc. Likewise preamp "outs" are in one row and mixers "in" are in the next.
3. Monitor amplifier "out" and speakers "in" should be located on the extreme right side of the jack field since this makes it possible to take the cables from such high level jacks off on that side of the rack without having them run parallel with other lower level cables from other jacks.
4. Miscellaneous jacks for spare equipment as transformers, pads, switches, etc. should be mounted below and out of the way of the important program carrying packs.
5. Use arrows to denote normalled through jack and multiples of jacks.

Plenty of use of the eraser will be found before the final satisfactory layout is made. *Figure 2* shows the final jack field for the studio layout being used for the purpose of discussion. It will be noticed that the above rules were not adhered to 100% for reasons of economy. A perfect jack field from the standpoint of the above rules would take more jack strips than actually necessary with a number of vacant jacks remaining.

Having completed the jack field, its position, along with amplifiers and other equipment to be mounted in the rack, can now be determined. A useful tool to aid in the layout of the rack is the outline shown in *Fig. 3*. This represents a standard cabinet rack having 77 inches of panel space. The scale is laid off

[*Continued on page* 41]

# Transition Frequency Compensation

C. G. McPROUD*

**The simple equalizer circuits described make it possible to correct any magnetic or crystal pickup for the various turnover frequencies used in recording.**

EVERY RECORD ENTHUSIAST has found certain records that sound fine with one system, but fail dismally on another. Yet another album may sound entirely out of balance on the first system, but reproduce perfectly on the second. Assuming that both systems are reasonably free from distortion and are comparable in other characteristics, this anomaly may cause some concern.

Since the recording characteristics of the various manufacturers are not yet standardized — and it seems doubtful that they ever will be — the reproducing equipment must be made sufficiently flexible to ensure satisfactory performance with all makes of records. At present, we are faced with transition or turnover frequencies of 500 and 800 cps on records of American manufacture, while the English Decca ffrr disks are reputed to be recorded with a 300-cps turnover. Obviously, no single equalization can compensate perfectly for these variations.

It is general practice for the recording characteristic to be essentially flat above the transition frequency. Below this point, a droop of 6 db per octave is normal, but this may vary somewhat in practice. The curve does not make a sharp bend at the turnover point, fortunately, on account of the simple means used to achieve the bend. Consequently, a simple means of equalization is sufficient for satisfactory correction.

Reproducing characteristics vary between magnetic and crystal pickups, so the type of equalization varies considerably. Both will be discussed here, with actual tested circuits.

*Managing Editor, AUDIO ENGINEERING

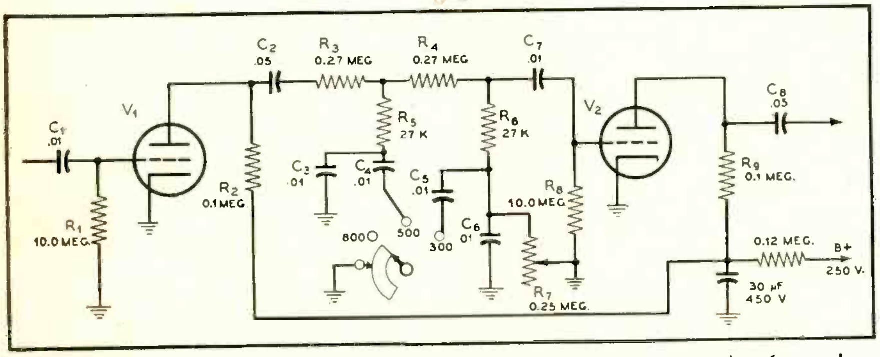
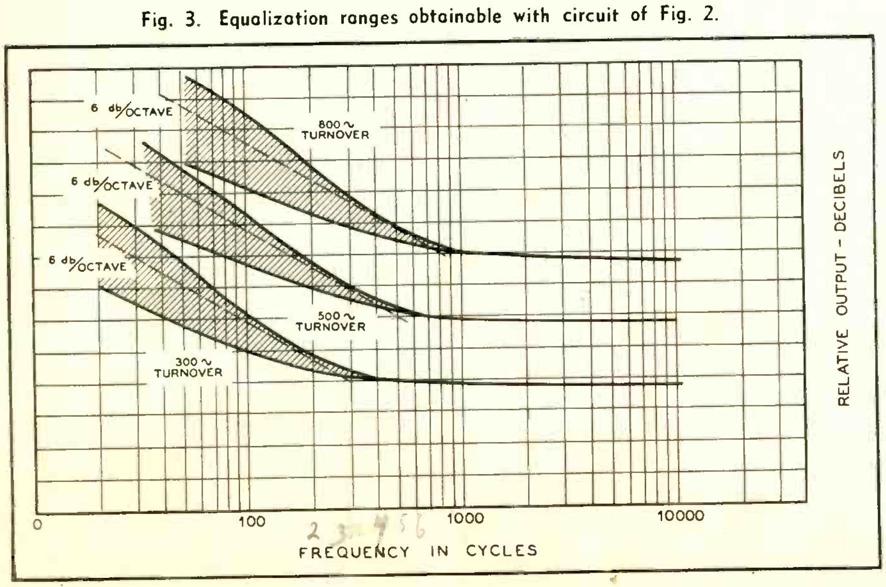



Fig. 2. Equalizer two-stage amplifier suitable for low-frequency compensation for modern high-quality, low-level magnetic pickup cartridges.

Fig. 3. Equalization ranges obtainable with circuit of Fig. 2.




## Compensation for Magnetic Pickups

The high-quality, low-level magnetic pickup, such as the GE and Pickering cartridges, require that some preamplification be supplied, and this provides a suitable place for the equalization.

A 6 db/octave boost requires at least 20 db of equalization for the entire range. When using *RC* circuits, a total loss (in the unequalized band) of 40 db is usually sustained to secure a net boost of 20 db, such as from 500 cps to 50 cps. If the output of the pickup cartridge is of the order of 50 millivolts, let us say, the loss due to equalization gives a *net* output voltage of 1/100th of this amount, or 0.5 mv. To get a 1-volt output signal to feed an ordinary amplifier requires a voltage gain of 2,000, or approximately 45 per stage for a two-stage preamplifier, indicating the use of high-mu triodes. The equalizer circuit may be inserted between them.

A single-section *RC* network, such as that of *Fig. 1(A)*, gives a rise of slightly over 3 db/octave at an insertion loss of 20 db. Putting two such circuits in series as at (*B*) doubles the boost, and doubles the insertion loss, of course. Thus, with proper choice of values for *R* and *C*, the required equalization can be obtained. The transition frequency is controlled by the size of the capacitors; the amount of equalization by the ratio between $R_2$ and $R_1+R_2$.

*Figure 2* shows a complete equalized amplifier suitable for use with a high-quality magnetic pickup. $V_1$ and $V_2$ may be the sections of a 6SL7, 6SC7, or 7F7; or they may be two separate high-mu triodes such as 6SF5's, or two

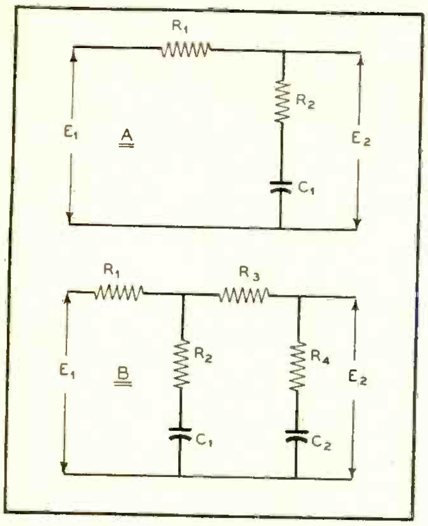



**Fig. 1. (A) Single-section RC network used to provide low frequency boost at the rate of 3 db/octave. (B) Two sections are necessary for a boost of 6 db/octave.**

duo-diode triodes such as 6SQ7's, or 6AQ6's, with the diode plates strapped to the cathodes. The values shown will give a total boost of approximately 7 db/octave, with the switch lowering the transition frequency by adding more capacitance.

Shunting the capacitance of one of the *RC* sections with the potentiometer $R_7$ provides a controllable boost ranging from 3 to 7 db/octave. This potentiometer has an audio taper for smoothest action, the connection being made so that clockwise rotation of the knob increases the resistance across the capacitors, and thus increases the bass boost. *Figure 3* shows the possible equalizations obtainable with this circuit.

The plate-load resistors $R_7$ and $R_9$ may be varied between 50,000 and 270,000 ohms to adjust the total gain of the preamplifier, and the output is intended to feed a 0.5 to 2.0 meg. volume control.

**Crystal Pickup Equalization**

The problems are entirely different when a crystal pickup is used. These devices normally give a flat output over the frequency range up to the transition point, and droop at the rate of 6 db/octave beyond. The low-frequency response is dependent somewhat on the total load into which the cartridge works, but if this load is greater than 1.5 megohms, the bass may be assumed to be normal.

Since a droop of 6 db/octave corresponds to a loss of 20 db for a frequency ratio of 10:1, the insertion loss must be at least 20 db to compensate for the range from 300 to 3,000 cps for the lowest turnover frequency. An additional 10 db would carry the compensation to 9,000 cps, which is a reasonable upper limit for this type of reproducer. However, most amplifiers used with crystal pickups do not provide sufficient gain to permit the use of this much equalization, so a circuit for a maximum of 20 db will be presented.

If the total insertion loss at low frequencies is to be 20 db, the resistors $R_1$ and $R_2$ of *Fig. 4* must provide it. Therefore,

$$I.\ L.\ (\text{db}) = 20 \log \frac{R_1 + R_2}{R_2}$$

Since $R_2$ may very well be a 0.25 meg potentiometer used for the volume control, and since $(R_1 + R_2)/R_2$ must equal $\log^{-1}$ 1.0, or 10.0, $R_1$ is 2.25 megohms.

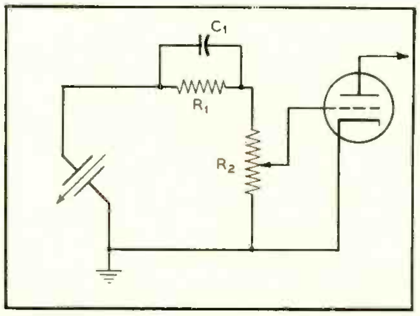



**Fig. 4. Type of equalizer used with crystal pickups to provide proper compensation for high-frequency droop inherent in capacitance generator.**

The capacitor across $R_1$ may be calculated for any desired transition frequency. The circuit of *Fig. 5* shows a complete equalizer, using a 1.0-meg potentiometer in series with a 1.2-meg resistor for $R_1$, thus permitting the control of the amount of high-frequency boost, while the switch $S_1$ controls the transition frequency. $S_2$ is used to control the low-frequency response, providing a variety of load resistances for the pickup to work into, with the attendant change in bass output.

**Conclusion**

These circuits may be useful to the experimenter, engineer, or record enthusiast for providing a wider range of control for record reproduction. Once the correct settings for the equalizer controls are determined for each individual record or album, they may be marked on the records to facilitate future use. Proper equalization will do more toward making the reproduction a realistic performance than any other single influence, when used with good quality equipment.

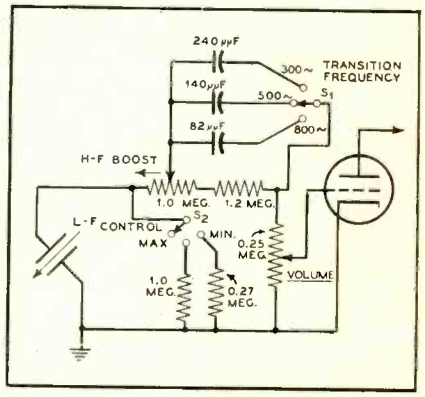
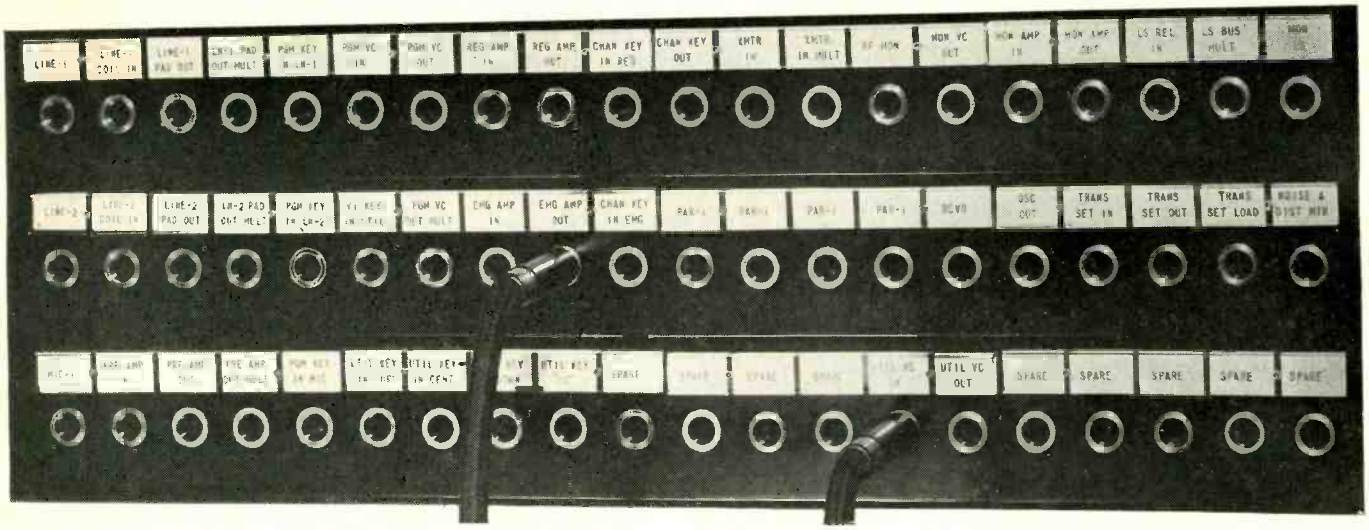



**Fig. 5. Complete equalizer circuit to be used between crystal pickup and grid of first tube, making use of both low- and high-frequency compensations.**

**Douglas Butler recording radio broadcast program in the control room of studio 5A at WNBC.**

*—NBC Photo*



# Single Jacks for Broadcast Application

**HOWARD A. CHINN* and ROBERT B. MONROE****

**Single jacks provide better performance, require less space, and cost less.**

A BANK OF TELEPHONE-TYPE JACKS, known as a jack field, is employed in essentially every broadcasting, recording, or public-address audio system. It is the usual practice to connect the input and output of the important circuits and components of the system to these jacks. The connections are made in a manner which permits access to the circuits by placing a plug or patch cord into the proper jacks. Furthermore, auxiliary "make" contacts are employed on each jack so that when there are no patch cords in the jack field, the components and circuits are connected together in their *normal* sequence, see *Fig. 1.* These auxiliary contacts are called "normals", for obvious reasons.

A jack field, incorporated in an audio system, provides a high degree of operational flexibility. By the use of patch cords various operations may be quickly performed, as for example:

(a) A defective component may be readily replaced by another similar unit, or may be removed completely from the circuit by patching around the unit.

(b) Whenever necessary, the program circuits may be rearranged as desired.

(c) Special-effect devices, such as filters or reverberation facilities, may be inserted in any desired program channel.

In addition to the operational conveniences made possible by the jack field, test and measurement of the facilities are greatly simplified as it is possible to make direct connection to the terminals of various components of the system. Location of a defective unit is thereby greatly facilitated.

The plugs currently employed in the majority of professional audio systems are the "twin" type, *Fig. 2.* With these twin-type plugs it becomes necessary to employ jacks in pairs. This arrangement of twin plugs and jack pairs has been in use in broadcast plants since the early days of radio broadcasting. Their introduction to radio work was a carry-over from the practice on "Long Line" circuits used for network program distribution at telephone exchanges. The performance of these plugs and jacks over a period of many years has always been reliable. However, the following points, which may be considered disadvantages, have been noted:

(a) Care must be exercised to insert the plug correctly polarized. If this is not done the polarity of the circuit may be reversed.*

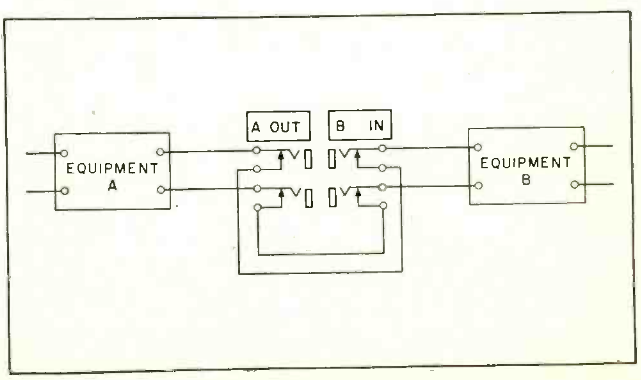



Fig. 1. Jacks incorporated in an audio system permit access to the input and output of the various equipment units by placing a plug or patch cord into the proper jacks. When no patch cords are in use the components are automatically connected in their normal sequence by means of the auxiliary jack contacts shown. These contacts and the associated wiring are called circuit "normals".

*This may lead to several undesirable consequences. In unbalanced-to-ground arrangements reversal of the plug will ground the wrong side of the circuit. If this does not short circuit the program material (depending on grounding arrangement employed) it will place a ground on the high side of unbalanced elements. In most cases, particularly in the case of filters and attenuators, the characteristics of the unit may be considerably altered. Polarity reversal in both balanced and unbalanced-to-ground arrangements causes loss of phasing, if employed.

**Chief Audio Engineer, Columbia Broadcasting System.*
***Engineer, Columbia Broadcasting System.*

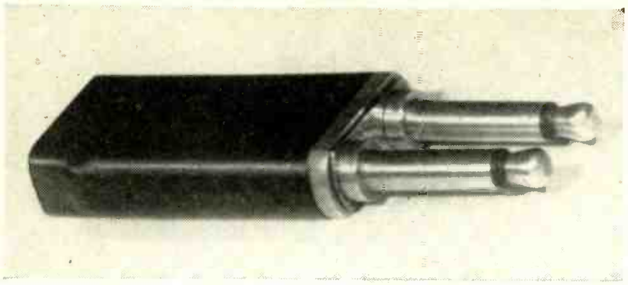


Fig. 2. The "twin" type of plug currently employed in the majority of audio systems. Jacks are used in pairs with this type of plug.

Fig. 3. A tip, ring, and sleeve type of single plug often used in telephone switchboard work. This type of plug is not satisfactory for use in audio systems as explained in text.

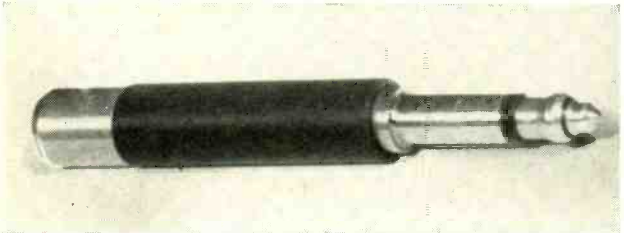


Fig. 4. This type of single plug, constructed with a shorter tip than the plug of Fig. 3, is satisfactory for use in audio systems. It is used in conjunction with the jack in Fig. 5.

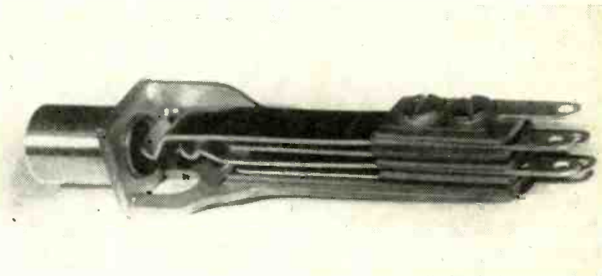


Fig. 5. The type of single jack shown above was used in all the measurements, investigations, and applications described in this text.

(b) A relatively large amount of panel space is required for the jack field.

(c) The cost of the jack field and associated patch cords is high.

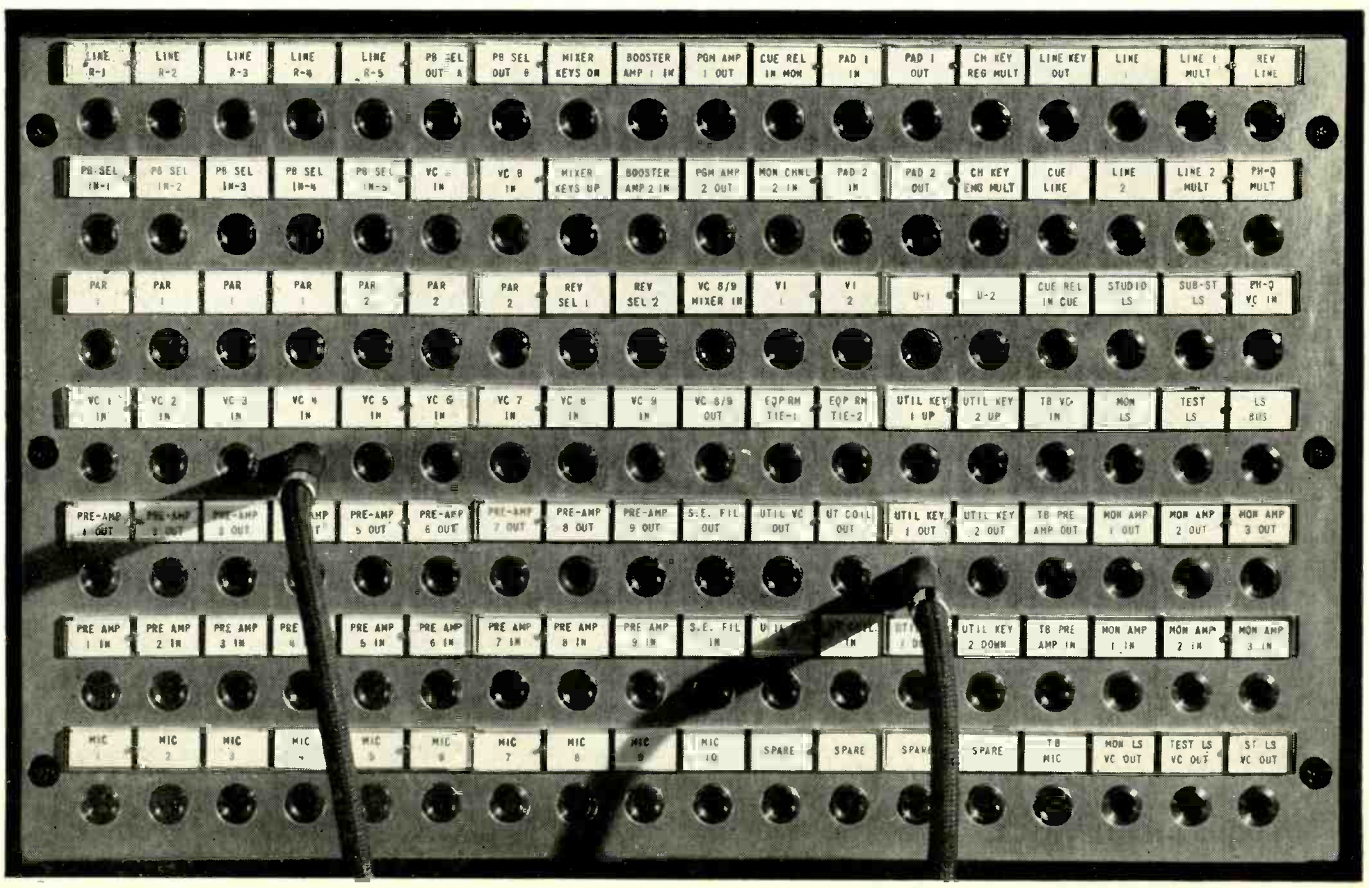
Inasmuch as the trend in the design of audio facilities is to reduce the size of installations, a study was made to determine the possibility of employing single plugs and jacks without sacrificing any of the features of the traditional twin units. The successful application of single jacks would overcome the shortcomings of the double units mentioned above; that is, less panel space would be required for a given number of jack circuits; the cost would be lower; and all possibility of reversing the circuit by improper insertion of the plug would be eliminated.

### Single Plugs and Jacks

To perform the same operations now accomplished by twin plugs, it is necessary that the single plug accommodate three conductors; two for the program circuit and one for ground. This requirement can be met by the "tip, ring and sleeve" type plug of the general type used on telephone switchboards.

It must be pointed out, however, that the usual telephone-type single plugs (such as the W.E. type 110) are *not* satisfactory for broadcast use, *Fig. 3.* This can be readily observed by visual inspection of the operation of these units. As this type of plug enters the jack, a point is reached where the tip of the plug momentarily touches both program springs of the jack. During this period, the plug short-circuits pro-

Fig. 6. The jack field of the CBS 3-B studio control console. 126 jack circuits are mounted on a panel only 9 by 14½ inches in size. The spacing between jacks is ¾ inches, the spacing between rows 1¼ inches.



gram material on the jack. For this reason, these telephone-switchboard plugs cannot be considered for broadcast use.

This difficulty can be avoided by employing a different type of tip, ring and sleeve plug constructed with a shorter tip. Such plugs have been made for telephone applications and are available. An example of a plug of this type is the W.E. type 291-A which was employed in this study, *Fig. 4*.

The single jacks employed were the W.E. type 239-A. These jacks are designed to operate with the W.E. type 291-A plug and contain the necessary contact springs for two program circuits, each of which is provided with a "normal" contact. The ground circuit is carried in the usual manner, through the frame of the jack which is equipped with a terminal lug, *Fig. 5*.

It is often necessary in the design of audio systems to employ a few jacks equipped with auxiliary "make" or "break" contacts to accomplish some special circuit changes when a plug is placed in the jack. Jacks of the type under discussion are also available with these auxiliary contacts.

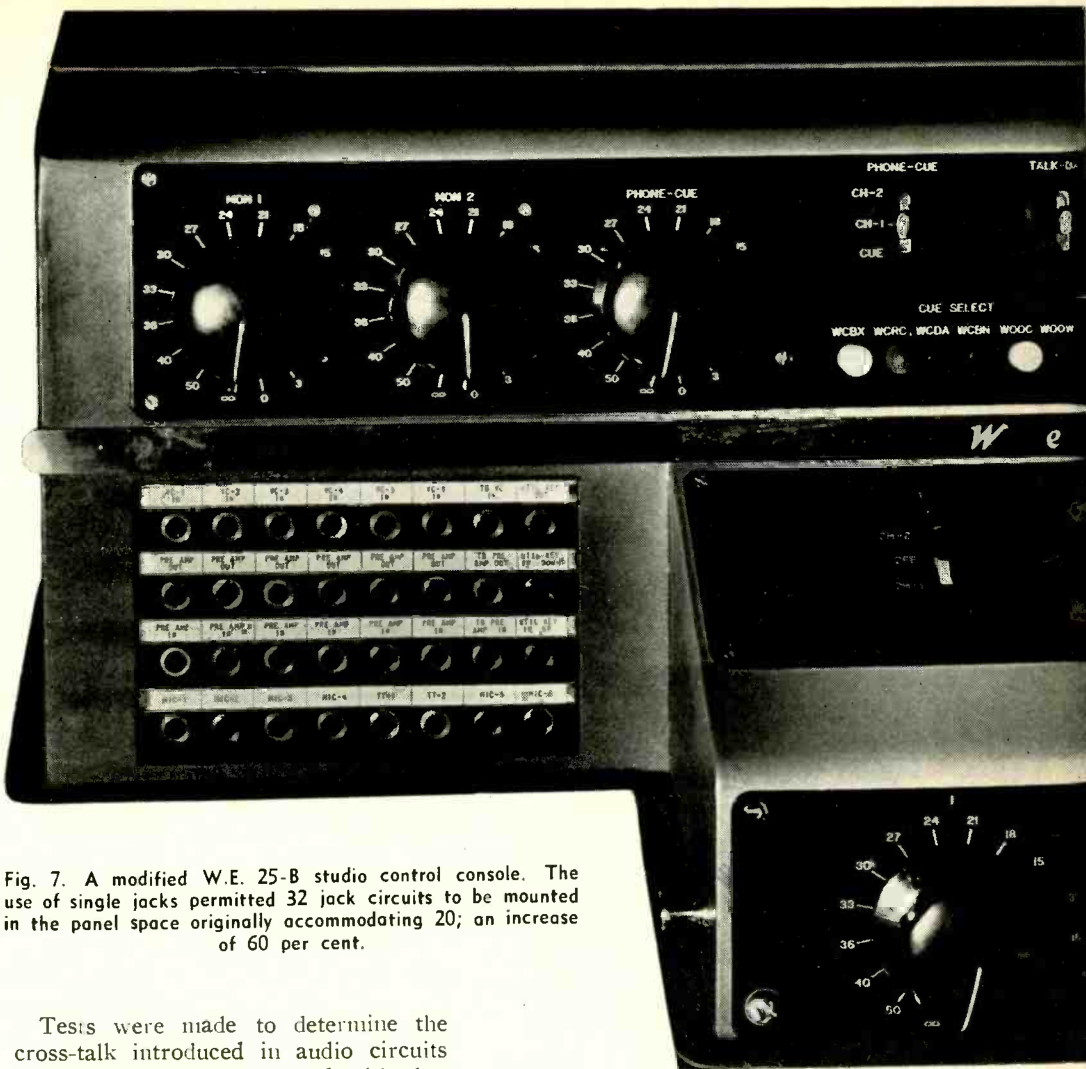



Fig. 7. A modified W.E. 25-B studio control console. The use of single jacks permitted 32 jack circuits to be mounted in the panel space originally accommodating 20; an increase of 60 per cent.

### Experimental Tests

As part of the early study of single plugs and jacks, an audio jack field consisting of 24 single jacks mounted on a conventional jack mounting panel was set up for the purpose of checking the operational performance of single plugs and jacks. Various amplifiers and other components were connected to this jack field, which was then normalled to simulate a conventional studio audio channel complete from microphone input circuits to program line output. Visual and aural monitoring facilities were included in this experimental set-up to make it resemble as closely as possible an actual installation.

The plugs and jacks in this experimental system were associated with both balanced-to-ground and unbalanced-to-ground circuits. In the case of the balanced circuits both the tip and the ring of the plugs, as well as the corresponding contacts of the jacks, were high with respect to ground; however, the wiring was so arranged that in the case of unbalanced circuits the tip was high and the ring low.

Patch cords, equipped with single plugs, were employed to check the ability of this jack field to perform all the usual jack field operational requirements, such as substituting amplifiers, patching out various components and making multiple connections.

It was found that the operation of these single jacks and plugs was entirely satisfactory and equivalent in all respects to the performance of the conventional double units.

Tests were made to determine the cross-talk introduced in audio circuits by single jacks as compared with that introduced by the more conventional "double" jacks. These tests were made on a balanced-to-ground, terminated, 600-ohm circuit by applying a sine wave of 15,000 cps to a jack, or in the case of double jacks to a pair of jacks, and measuring the signal crossing into adjacent jacks. The results of these tests are tabulated below:

| Cross-Talk To: | Single Jacks | Double Jacks |
|---|---|---|
| Adjacent Left-Hand Jack(s) | —139 db | —132 db |
| Adjacent Right-Hand Jack(s) | —137 db | —128 db |
| Jack(s) Immediately Above | —140 db | —114 db |

It will be noted that cross-talk is lower in the case of single jacks. *This is an important consideration in the design of audio facilities for present-day standards.*

### Shunt Capacity

Inasmuch as shunt capacity is an important factor in audio systems, measurements were made of the shunt capacity of both single jacks and pairs of jacks. The measured values are as follows:

SHUNT CAPACITY IN MICROMICROFARADS

| | |
|---|---|
| **Single Jacks** | |
| Tip-Spring to Ring-Spring | 10 |
| Tip-Spring to Ground | 14 |
| Ring-Spring to Ground | 10 |
| **Double Jacks** | |
| Tip-Spring to Tip-Spring | 5.5 |
| Jack 1 Tip-Spring to Ground | 13 |
| Jack 2 Tip-Spring to Ground | 13 |

It is seen that the shunt capacity between program springs is slightly higher in the case of single jacks. It is not believed, however, that this will cause any difficulty in new audio facilities, inasmuch as 150-ohm circuits will probably be used in most future systems designed for wide-band transmission. The reduction of circuit impedance by a factor of four will more than offset any effect of the slightly greater shunt capacity on the higher audio frequencies.

### Practical Application

Audio installations employing single plugs and jacks have now been made at CBS stations in New York, St. Louis, and, presently, in Hollywood. Three of these installations are shown in *Figs. 6, 7*, and *8*. In addition, Station WWL at New Orleans has incorporated these jacks in a post-war studio control console.

As a result of the experience which has been gained with these installations, it has become evident that single jacks offer several advantages over the twin units. These advantages include: (a) smaller space requirement, (b) lower cost, (c) lower cross-talk, and (d) impossibility of incorrect patch cord insertion.

It is believed that the use of single jacks of the type that retain all the advantages of double units marks a step forward and will find wide application in the future.

# Preamplifier Noise in FM Broadcasting

**A. E. RICHMOND***

**Methods of testing and improving the signal-to-noise ratio in preamplifiers are discussed. These methods are applicable to all sound systems.**

The Federal Communications Commission has established stringent standards concerning the performance of frequency modulation broadcast stations. These requirements must be satisfied prior to the granting of a license for regular operation. It is the purpose of this article to approach in a practical way one aspect of a problem presented by these requirements[1] — *the necessity of limiting the noise to a value not greater than 60 db below 100% modulation.* (This level corresponds to a signal-to-noise ratio of 60 db, or 1,000,000 to 1.)

In the application for a construction permit for an FM broadcast station, a statement is included that, in applying for license to cover the construction permit, the applicant will be required to submit measurements to show that the transmission system will actually comply with these standards. There is also the possibility of such measurements being made by FCC representatives during inspection of the station.

The 60 db limit to the noise is specified as being applied to the entire system, from the input terminals of the microphone preamplifier to the transmitter output. However, the contribution of the preamplifier noise to the total noise of the system is of major importance, and this article will be limited to consideration of the preamplifier noise in the FM broadcasting system.

Chinn[2] has described mixing-circuit requirements for minimum noise, and has shown that *the signal-to-noise ratio of a properly designed broadcast system will, in general, be defined by the signal-to-noise ratio existing in the input stage of the microphone preamplifier.* Hence, the importance attached to preamplifier noise.

Before discussing any specific method to be employed in obtaining noise level measurement, a few facts concerning preamplifier noise will be considered. Because of the low output volume level of available wide-range microphones, it is necessary to amplify the microphone output by means of a preamplifier before subjecting the audio signal to attenuation in a volume control pad (high level mixing). Even with this precaution, considerable attention to preamplifier noise is necessary in order to get this noise down to 60 db below the approximate —60 vu output of a broadcast microphone. The noise at the preamplifier input stage cannot be greater than about 120 db below one milliwatt. After preamplification, the signal must further be amplified at any point in the circuit where it has been reduced dangerously near a —60 vu level by losses in mixing controls or master volume controls, or in mixing networks.

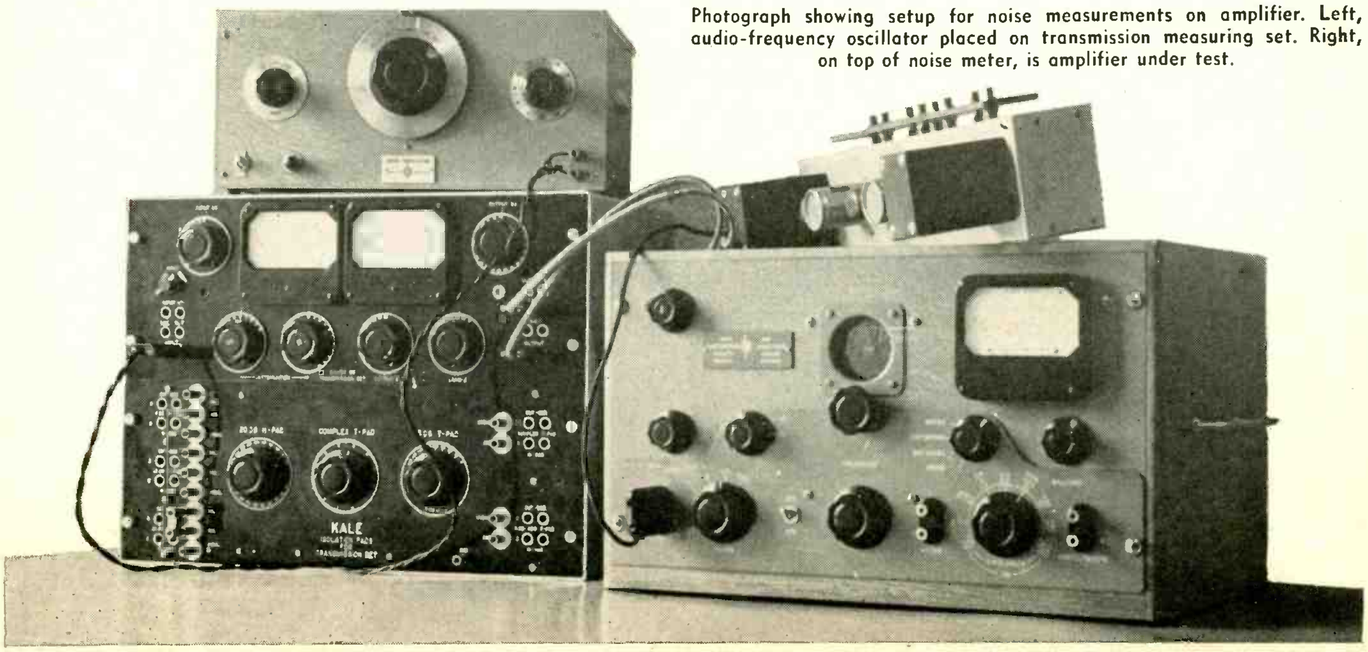
## Specifications

It is obvious that not just any "good" preamplifier will satisfy the requirements stipulated for FM. Unfortunately, not all specifications for preamplifiers are precise with regard to noise levels to be expected in practice. Often one finds amplifiers specified as to (*a*) gain in db, (*b*) maximum output level in vu or in dbm for small distortion, and (*c*) output noise level in db below the specified maximum output level or some

*Chief Engineer, KALE-KALE-FM*

[1] Federal Communications Commission, "Standards of Good Engineering Practice Concerning FM Broadcast Stations," Effective Sept. 20, 1945; Revised to Jan. 20, 1946 (U. S. Government Printing Office, Washington 25, D. C.; 10¢), page 14.

[2] H. A. Chinn, "Broadcast Studio Audio-Frequency System Design," *Proc. I.R.E.*, Vol. 27, page 86, Feb., 1939.




Photograph showing setup for noise measurements on amplifier. Left, audio-frequency oscillator placed on transmission measuring set. Right, on top of noise meter, is amplifier under test.

other stated output level. Regrettably, the picture to be gained from such noise level specifications requires proper interpretation to avoid confusion. *The output level available from the microphone will not necessarily be sufficient to drive the microphone preamplifier to the specified amount and thus obtain the stated signal-to-noise ratio.*

As an example, assume a microphone amplifier having a gain of 40 db, a maximum rated output of +20 vu, and a signal-to-noise ratio of 80 db at the rated full output. It will be seen that the noise at the output terminals is $20 - 80 = -60$ vu. The signal level at the output, if we assume a microphone level of $-60$ vu, will be $-60 + 40 = -20$ vu. The noise will thus appear to be only 40 db below the signal, and measurements will probably show the station performance to be sub-standard as regards FCC requirements.

It is not the purpose of the present article to offer detailed information on methods of amplifier noise reduction, which are well discussed in the literature. However, some considerations which have been found important will be mentioned. Preamplifier noise causes include (1) thermal agitation, (2) microphonics, (3) hum and (4) tube noise. Thermal agitation causes noise apparently irreducible[2] below about $-130$ vu. Preamplifier design precautions should include shock mounting for the amplifier to combat microphonics. Hum can be counteracted by such precautions as (1) mounting the amplifier out of the way of alternating magnetic fields, (2) constructing the chassis of non-magnetic material,[3] and (3) shielding the input transformer with alternate iron (or permalloy) and copper shields.[4]

Tube noise will prove to be a problem in getting the over-all noise sufficiently low to meet requirements. While the high gain of a pentode, beam, or screen-grid tube would be attractive in the input stage, the noise level must be expected to be higher (for the same gain) than as if a triode were used.[5] A tube recommended to the writer for use in preamplifier input stages, and which shows promise, is the 7F7 twin triode, connected in push-pull. The high gain and low noise level of this type lend themselves well to preamplifier service.

Incidentally, the use of negative feedback, while of benefit for some purposes, will show no improvement (and no damaging effects) with regard to the input signal-to-noise ratio,[6] at least as a direct result of the feedback. This fact is because of the limited available input to the amplifier from the microphone. The reduction in noise will be matched by a similar reduction in signal, leaving the ratio at the input stage (and at the output) unchanged.

In actually measuring the noise performance of a complete FM installation, the writer is informed by the FCC that the intent is to determine the noise level with gain controls at a normal setting. (While this intent is good in that it attempts to get at a figure representative of actual operating noise, it would seem to the writer that it leaves some undesirable latitude for interpretation. Possibly a quantitative input-circuit performance requirement could be devised, with provision in the Standards of Good Engineering Practice for special consideration if an individual applicant had an unusual microphone type with higher output than is usually associated with microphones of acceptable fidelity.)

The noise level at the output of the transmitter (after FM detection and de-emphasis) must be determined to satisfy the FCC performance requirements, but measurements may be made quite simply on the preamplifier itself to determine what limitation, if any, will be placed by the preamplifier on the over-all noise level. In order to obtain readings indicative of preamplifier performance in the complete system, the noise indicator should be the same as, or similar to, the instrument to be used in measurements to prove the performance of the entire transmission system. The noise meter itself is essentially a sensitive audio-frequency voltmeter, responding to the range of 50 to 15,000 cycles. No such regularly available instruments known to the writer include the FCC-required feature of an indicating instrument having ballistic characteristics similar to those of the standard vu meter.[7] The Hewlett-Packard Model 330B Distortion Analyzer may be modified on special order to the "Specification 11296" Distortion Analyzer, with the required vu meter characteristics.

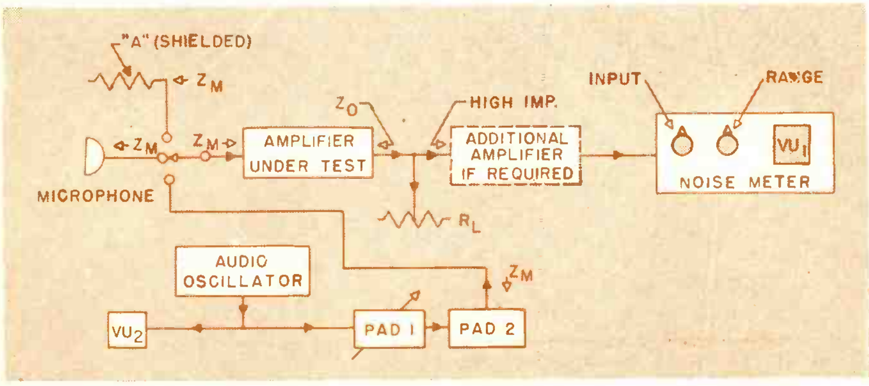



**Fig. 1. Diagram of connections for making measurements of preamplifier noise relative to program level obtained from microphone. Pad 1, Pad 2, meter VU 2, $R_L$, and in some cases the audio oscillator, may be found in a transmission measuring set.**

## Noise Measurements

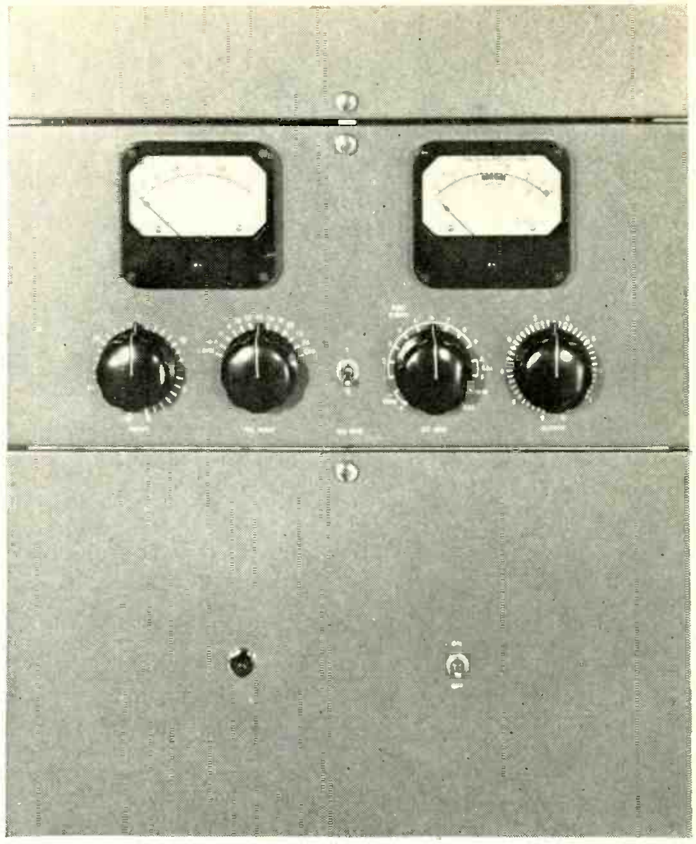
The simple hookup shown in *Fig. 1* is suggested for making noise measurements on FM preamplifiers. The amplifier is operated into its rated load resistance, $R_L$, as well as being connected to the high-impedance input terminals of the noise meter. The range multiplier switch on the noise meter is first set on one of the higher ranges, and the input of the preamplifier under test is connected to the microphone. The microphone is then actuated by actual program material typical of that with which the preamplifier will be used. The input control on the noise meter is then advanced until a reading of 0 db is obtained on the volume indicator. The shielded impedance *A* (equal to the internal microphone impedance) is now substituted for the microphone, and the range switch of the noise meter is set so that the preamplifier noise causes a meter deflection of about 0 db, without the previous input control setting being disturbed. Noise level below 100% modulation, as limited by the particular preamplifier and microphone tested, is indicated by the number of db through which the range multiplier switch was turned, *plus* any final reading of the instrument pointer which is *less* than 0 db, or *minus* any reading which is *greater* than 0 db.

The noise reading will not necessarily be steady, especially if the preamplifier input tube has some "flicker effect",[8] and if the hum has been well reduced. It is suggested that the standard method of reading the vu meter[9] be employed in determining the levels of both pro-

[*Continued on page* 45]

[2] F. E. Terman, "Radio Engineers' Handbook," page 478.
[4] Reference 3, page 132.
[5] Reference 3, page 294.
[6] F. Langford Smith, "Radiotron Designer's Handbook," Third Edition, page 37.
[7] Reference 1, page 15.
[8] Reference 3, page 292.
[9] American Standard: "Volume Measurement of Electrical Speech and Program Waves" (American Standards Association, 29-33 West 39 Street, New York; 20¢), page 7.

# Performance and Use of Limiting Amplifiers




**W. W. DEAN* and L. M. LEEDS***


**The limiting amplifier shown at right possesses special features which provide improved performance**

RECENT ADVANCES in limiting amplifier design permit a substantial increase in average modulation level of a-m stations without overmodulation, and without the effects of limiting becoming noticeable to the listener. These improvements also help to prevent overswing in connection with f-m transmitters. The General Electric Type BA-5-A Limiting Amplifier incorporates certain circuit features, based on developments by Columbia Broadcasting System engineers, which make this improved performance possible.

Earlier types of limiting amplifiers permitted a signal strength increase of from 3 to 5 db—but with considerable overmodulation on sharp peaks and a limiting action which was audible on certain types of programs.

### Special Features

This new limiting amplifier will permit a much higher degree of limiting without any audible evidence of limiting action and with substantially no overmodulation, even on sharp peaks. This performance is obtained through the use of a very high compression ratio, a very short attack time, and an automatic variation of recovery time.

The ideal limiting amplifier is one in which the output increases directly as the input, up to a specified point known as threshold, and does not increase with further increase in input. This ideal limiter would accept signals whose peak intensity just equalled the threshold level, without change in the normal gain relationship. Input signals above threshold value would instantaneously cause a reduction in db gain exactly equal to the db increase in input level, thus holding output constant.

Limiting amplifiers previously available have departed from the ideal by allowing an appreciable increase in output signal with inputs beyond threshold, and by not having gain reducing actions rapid enough to catch fast peaks.

A measure of the limiting action is the "compression ratio". This is the ratio of the change in input level (in db) to the change in output level (in db) for a stated increase in input level above threshold. Thus the compression ratio of a limiting amplifier tells how effectively it holds the output constant when the peak input signal exceeds the threshold value.

*Figure 1* shows that the new amplifier has an effective compression ratio (expressed in db) of 24 to 1 over the range from threshold to a gain reduction of 12 db. The variation of output level over this range is only 0.5 db. This is believed to be a considerable improvement over limiting amplifiers of earlier design.

Although an adequate compression ratio, or gain reduction characteristic, is essential it is also necessary that the compression characteristic be effective


**Transmitter Div., Electronics Dept. General Electric Co. Syracuse, N. Y.*

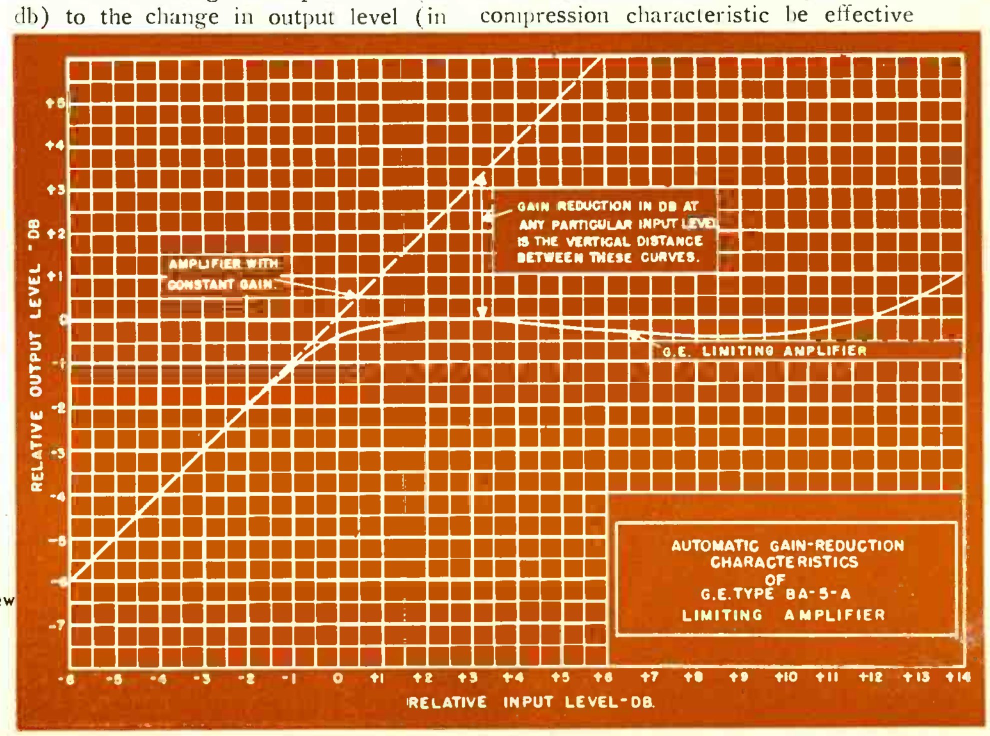




Fig. 1. Compression characteristics of the new amplifier.

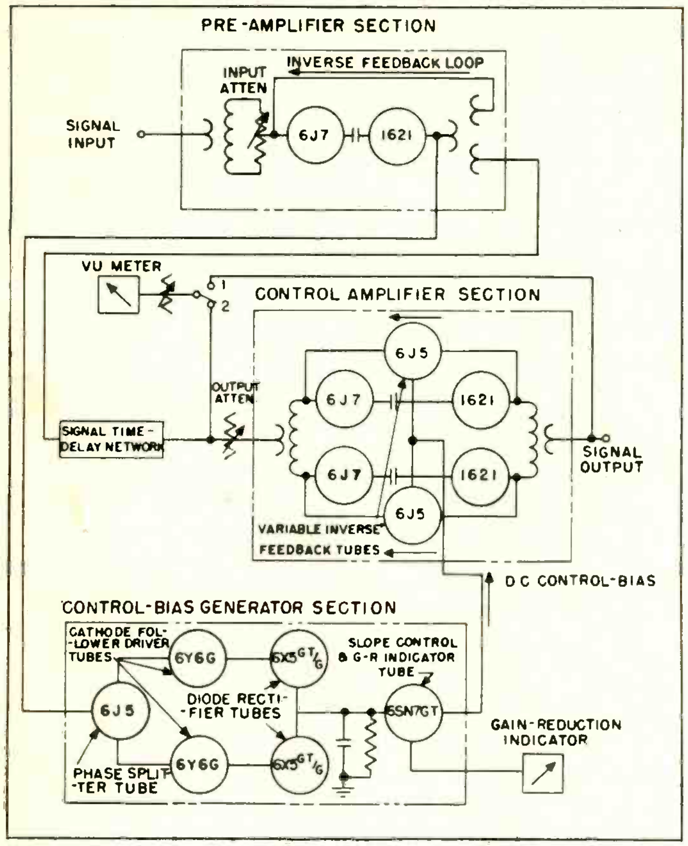



Fig. 2. Bias generator used to control gain.

for rapidly changing waveforms as well as for those changing slowly. If the limiter is to catch fast program peaks, voltages rising in a matter of microseconds must be controlled in accordance with the curve. Otherwise severe instantaneous overmodulation would result. Thus a good limiting amplifier must be very fast-acting; that is, it must have a short attack time.

An action sufficiently rapid for fast program peaks is obtained in this limiting amplifier through the use of a special low impedance, high-speed bias generator which is used to control the gain of the controlled amplifier, *Fig. 2.* Although the bias generator itself is fast-acting once the signal arrives at the control element, there is a small time delay associated with the arrival of the signal. In order to obtain superior performance, a compensating time-delay network is located just ahead of the controlled amplifier which operates to delay the signal in the main channel for just that time necessary for gain reduction to be effectuated. Thus the unit anticipates peaks and effectively prevents overshoot.

### Recovery Time

The gain-recovery time of a limiting amplifier must be properly proportioned and preferably automatically adjusted in accordance with the number and duration of the peaks if completely satisfactory operation is to be obtained. Most limiting amplifiers have the recovery time adjustable in several steps. A compromise setting is usually sought in which the recovery time is made short enough so that one sudden peak won't depress the gain for too long a time, with the attendant "hole in the program," and yet long enough so that a series of spaced peaks won't cause a rapid shifting of gain or pumping. Pumping is especially noticeable when a series of words is spoken against a crowd or applause background. The up-and-down effect of the gain reduction on the steady background is most annoying.

The new amplifier has a unique circuit that automatically varies the recovery time in accordance with the nature of the program material. It is essentially a slow-charging and slow-discharging recovery circuit superimposed on a relatively fast recovery-time circuit. On a single peak this superimposed circuit does not charge sufficiently to affect the normal 0.5 second recovery time. However, on a series of peaks it has time to charge appreciably and thus limits the amount of gain recovery between peaks.

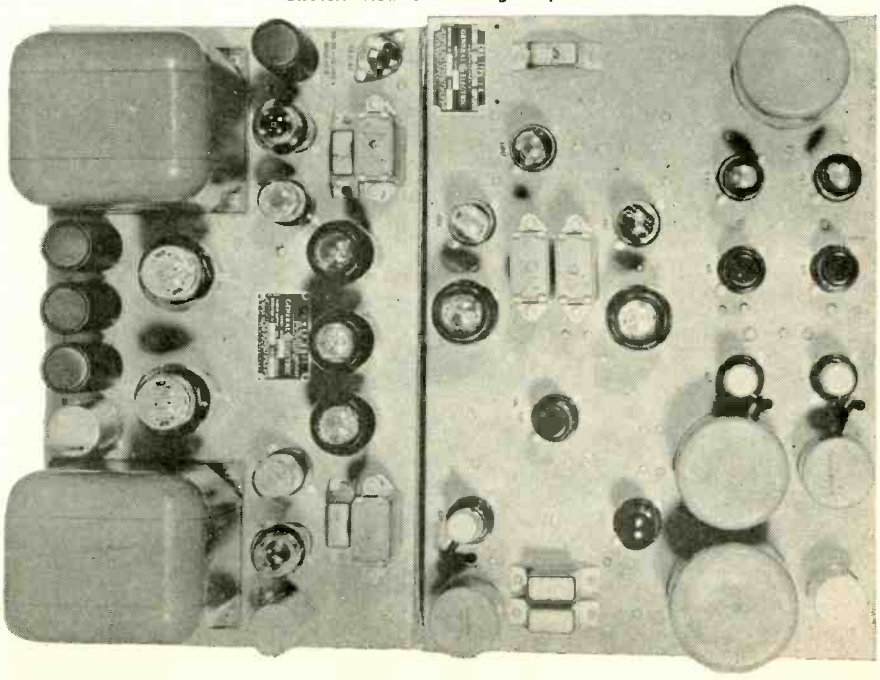
A flat frequency response and a low value of steady state distortion are necessary for a limiting amplifier to be used in broadcast service. The specifications should include performance under varying amounts of gain reduction as well as below threshold. *Figure 3* shows the typical frequency response of the new amplifier taken at 4 db below and at 10 db above threshold. It will be observed that the frequency response shown at these two widely different levels is within ±0.5 db which is well within the guaranteed maximum variation of ±1 db.

*Figure 4* shows typical total harmonic distortion at 12 dbm output under the same conditions. It will be observed that the distortion at these two widely different levels is well within the guaranteed performance.

### Limiter Adjustment

The operating adjustment of this new limiter in an a-m system is relatively simple and comprises the checking of the limiter itself, the setting of the output level so that 100% transmitter

Chassis view of limiting amplifier.



modulation occurs at the limiter threshold, and the advancement of the input level by the number of db of limiting desired.

Proper operation of the limiter itself is obtained through an adjustment of the slope or flatness of the compression characteristic, and d-c balance of the push-pull feedback tubes over their operating gain reduction range. Controls to handle these adjustments, which would be done normally only at infrequent maintenance periods, are located behind a small section of the hinged front panel. They are then out of the way so they will not be disturbed from day to day, yet are instantly available when wanted.

After checking the adjustment of the limiter characteristic, the next step is to adjust the output level so that the flat portion of the compression curve occurs at the 100% modulation point of the transmitter. With this limiter, there is substantially no change in output within a 12 db range in the limiting region. Consequently, the output can be set precisely at the 100% modulation point of the associated transmitter. This is done by adjustment of an output control with 45 positions in steps of 0.2 db. The nominal output level of the limiter is ±12 dbm which is ample to meet the input level requirement of broadcast transmitters which has been specified by the RMA to be 10 dbm ±2 db for 100% modulation. The output is thus designed to feed a 600/150 ohm resistive transmitter input directly without the necessity of intermediate pads.

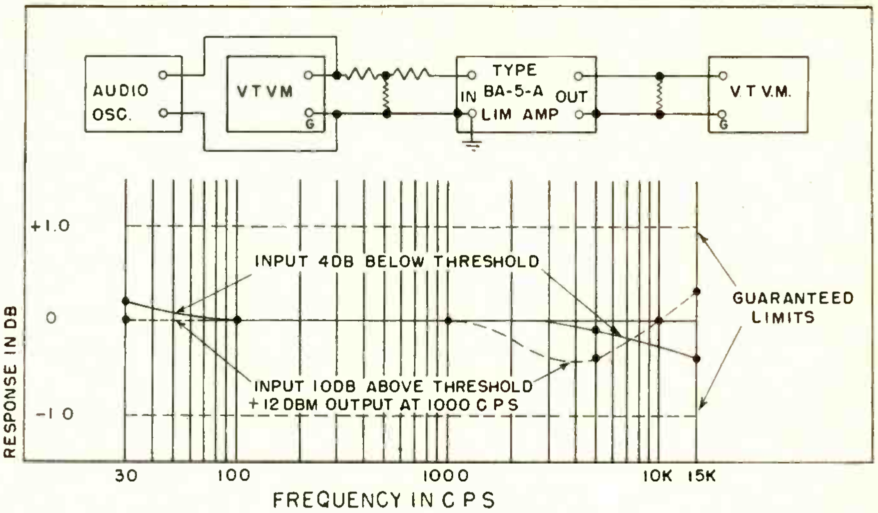



Fig. 3. Typical frequency response of the new amplifier.

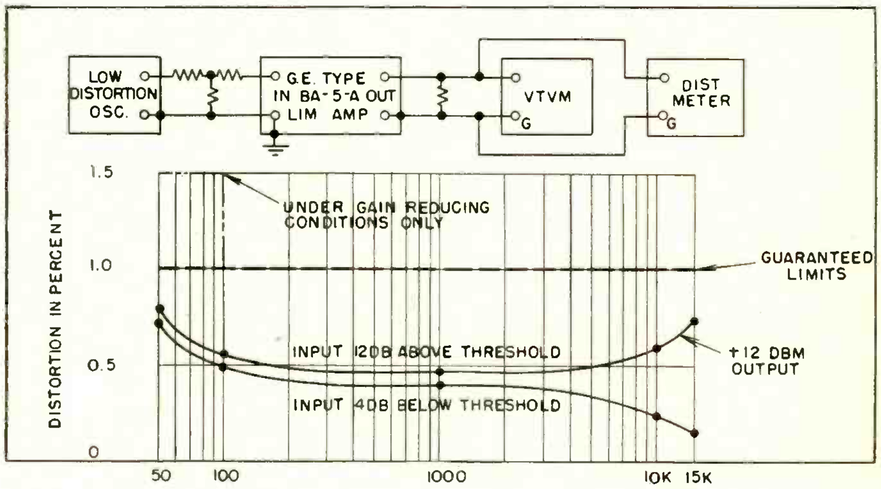



Fig. 4. Total harmonic distortion at 12 dbm output.

### Output Level Adjustment

The actual adjustment of the limited output level to the 100% transmitter modulation level is accomplished by applying a sine-wave input to the limiting amplifier, adjusting the input level until 3 to 4 db of gain reduction is observed on the meter, and then advancing the output level control until 100% modulation is obtained.

With the limited output level set to the 100% modulation point, the program may be applied to the input terminals. With normal program levels as customarily observed on the station VU meters, the input control may be advanced gradually until the gain-reduction meter kicks indicate occasional reductions of about 1 db. This means that program peaks are just above the threshold point. Now the input control may be advanced in 1 db steps until the desired amount of gain-reduction is reached. Our experience indicates that this increase may be as much as 8 to 10 db. Inasmuch as the original kicks indicated about 1 db reduction on the meter, the peak gain-reductions were at least 1 db; probably 2 or 3 db because of the inability of the meter to indicate the peak value of very short impulses. Therefore, to stay within the 12 db peak limiting range, the maximum number of db that the input control can be advanced beyond threshold is in the order of 8 to 10.

It must be remembered that the compression curve again turns upward about 12 db beyond threshold. Therefore, if all the limiting range is used to increase the program level, there will be no protection against occasional overmodulation should the program peaks exceed those that were prevailing when setting the original threshold level. It is therefore recommended, in the interests of good operating practice, to confine the gain-reduction beyond threshold to about 7 db. This actually amounts to some 8 or 10 db of instantaneous limiting on peaks.

When using a limiting amplifier in a pre-emphasized system, such as f.m., different operating conditions prevail. These new operating conditions necessitate a consideration of where to locate the pre-emphasis in the circuit and whether the limiting amplifier is to be used primarily as a device to increase average modulation, as in a.m., or as an overmodulation guard.

In f-m systems it has been observed that with most receivers distortion becomes distinctly objectionable if the transmitter swing is appreciably more than ±75 kc at any time. This is due to the relatively narrow receiver discriminator characteristic and the fact that a reasonably high quality audio system following the detector will not only pass any generated distortion directly to the listener but will also tend to bring it more forcibly to the listener's attention. Therefore, it is apparent that the maximum transmitter input voltage at all frequencies should not exceed the value that would produce ±75 kc swing.

In order to prevent overswing, the limiting amplifier should be located after the pre-emphasis network. In this way high voltages at the higher frequencies, which frequently result when certain types of program material rich in highs are passed through the pre-emphasis network, will be effectively limited in

[*Continued on page* 40]

To test the design let it be assumed that the grid resistors of the following stage are only 1000 ohms. Then $R_{L1}' =$ 978 ohms and $R_{L2}' = 980$ ohms. β can now be calculated from equation 9. β = 1.048. This result indicates that the design meets the requirement of a maximum 5 per cent deviation in balance, due to shunting of the output circuit. From *Fig. 1* the unbalance due to a 10 per cent change in $g_{m2}$ is found to be 0.9 per cent. $g_{m2}$ is taken as 682 μmhos from the first example.

**Power Output Stage**

With 6L6 tubes in Class A operation, let it be decided that about 60 volts drop across $R_K$ can be tolerated. Then from the tube data:

$E_p = 270V$
$E_{c2} = 270V$
$E_{c1} = -18V$
$I_K$ (total) = 145 ma d.c.
$R_K = 420$ ohms
$R_p = 23{,}500$ ohms
$Gmo = 5700$ μmhos
$N_s = 150$ } assigned
$R_L = 8$ ohms } assigned
$R_{pp} = 5000$ ohms } assigned

With these values $R_{L2}$ may be calculated from equation 33. In doing this, however, it must be remembered that $gm_2$ is an effective value. Hence $R_{L2}'$ is first determined using $gmo$. Then $gm_2$ is estimated using $R_{L2}'$ in equation 8. With this value of $gm_2$ a new value of $R_{L2}$ is found. In this case the second approximation will be close enough, but in others the designer may want to repeat the process until two successive values of $R_{L2}$ are equal or differ by a negligible amount. With two approximations $R_{L2}$ is found to be 2980 ohms. $R_{L1}$ is then 2020 ohms. From equation 39, $N_1 =$ 1515 turns and from 37, $N_2 = 2235$ turns. Except for mechanical construction and core material the transformer design is now complete.

**Power Output**

In Part I it was pointed out that when $R_{L1}$ and $R_{L2}$ are unbalanced, the grid voltage on $V_1$ will be larger than on $V_2$. For this example, then, it will be necessary to decide the maximum peak voltage allowable on the grid of $V_1$ in order to determine the corresponding power output. Let $eg_1$ peak be 16 volts. The input voltage is

$$e_s = eg_1 + R_K(i_1 - i_2) = \frac{eg_1}{1 - \frac{1}{1+R_K(gm_1+gm_2)}} \quad (41)$$

Substituting the values of $gm_1$ and $gm_2$ found by equation 8, and the value of $eg_1$ decided upon in equation 41, $e_s$ is found to be 27.3 volts peak. The r-m-s output power is

$$P_o = \frac{E_{pp}^2}{R_{pp}} = \frac{(2 \times V.G. \times .707 e_s)^2}{R_{pp}} \quad (42)$$

$$= \frac{2e_s^2(gm_1gm_2R_KR_{L2})^2}{R_{pp}[1+R_K(gm_1+gm_2)]^2} \text{ watts}$$

where $e_s$ is in peak volts. The power delivered to the load will be slightly less due to the transformer losses.

For the design under consideration with $e_s = 27.3$v, $P_o = 12.2$ watts. This is less than would be obtained with the same operating voltages in a conventional push-pull circuit, because the driving voltage on $V_2$ is less than on $V_1$. The power output could be increased by increasing $R_K$ but as pointed out previously there is a practical limitation due to the excessive voltage drop. A power gain of about 42 db is obtained with this circuit.

The voltage drop across $R_K$ is 61 volts for the quiescent condition. It is necessary therefore to supply a positive grid return of +43 volts to make the net bias −18 volts. Since $eg_2$ is developed across $R_K$, there is some loss of signal power in $R_K$. This amounts to about .15 watts at maximum output, which is small compared to 12.2 watts. The balance stability, as read from the curves of *Fig. 1*, is 3.2 per cent change in balance for 10 per cent change in $gm_2$. The voltage $eg_2$ can be found from

$$eg_2 = R_K(i_1 - i_2) = \frac{e_s gm_1 R_K}{[1+R_K(gm_1+gm_2)]} \quad (43)$$

In comparing this type of output stage with a balanced push-pull stage, it must be remembered that the distortions generated in the two sides of the circuit are not necessarily equal. This is because the grid voltages and plate loads are unbalanced. However, the higher grid voltage and lower plate load are on the $V_1$ side where there is negative feedback. It might be assumed, then, that this improves the cancellation of even order distortion in the output and that for some value of $R_K$ and a given pair of tubes, the circuit approaches true push-pull operation. The author has made no investigation along these lines, so the conclusion can be accepted only on the basis of its logic.

The cathode phase inverter output

[*Continued on page 47*]

Rex Morrow, assistant chief engineer at KUSC, seated at Western Electric 40A console.

# High Power Ultrasonics

**S. YOUNG WHITE**

**The inventor continues his discussion of ultrasonic equipment design and its practical applications.**

There are three natural divisions in ultrasonic-wave generation and application — gas, liquids and solids. Liquids and solids are roughly similar in impedance but gas is very much out of line with the others. In this article we shall consider only gas.

Sonic energy in gas is marked by very large values of acceleration and motion, and very low values of pressure in the wave, compared to liquids. It is true that a few watts of energy can be put into air from a quartz or magnetostriction unit by virtue of the very large resonant rise of such a practically unloaded unit, but in general we shall find it best to consider gas-type generators for gas loads, as they are a natural impedance match.

### Hartmann Whistle

One of the best known and simplest gas generators is the Hartmann, invented in Germany about 1927. This is a whistle of simple construction but rather complex theory. Since many of us have a half horse-power compressor in the shop, we can make one up rather easily, as shown in *Fig. 1.* Since there will be water particles and dust in the air supply, it is best to make it of hardened tool steel to resist abrasion, although for a short experiment it can be of brass or dural. For 40 pounds of air, the jet body can be drilled out with a #12 drill with an angle of 100 degrees, and a very small chisel point ground on it. Then it should be faced off carefully until the hole is about the size of a #48 drill. The edges should be quite sharp.

The cavity is drilled with a #48 drill and turned down to about a 90-degree angle. With an adjustable rod in the cavity, the device can be tuned through a range of from one to about 8 diameters deep. Then the units are mounted on the same axis, and provided with a longitudinal adjustment from one-eighth to one-half inch or so separation between units.

The 100-degree angle causes the air from the jet to actually cross over itself several times. The second cross-over point provides a negative resistance slope, and by placing our tuned cavity oscillation results.

It is a rather peculiar feature of any gas emitted from any orifice that as the pressure is increased, the velocity of the gas increases until it reaches sonic velocity, and then it cannot be increased by additional pressure. A rather simple way to look at this phenomenon is that a gas particle that has left the orifice at less than sonic velocity can send a message upstream that conditions are such

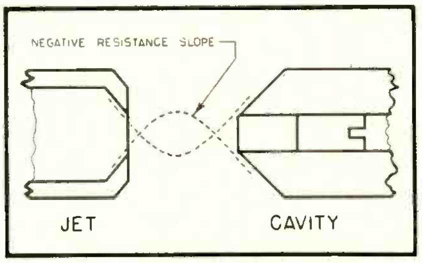



Fig. 1. Hartmann whistle design.

that more can come out. When sonic velocity is reached, however, the message "stays put" in the stream, and he cannot call out more of his fellows. When stated mathematically a confusing picture results, as some terms rise to infinity and others drop to zero.

### Sonic Velocity

In air at room pressure, sonic velocity is reached at 13.6 pounds pressure and the Hartmann whistle must operate above this point, so do not try to blow it yourself. The air packs into the cavity and blows out or explodes out in a series of diamond-shaped slugs which are quite easy to photograph. So the wave form is poor, and the frequency stability is also bad. Most of the air is unmodulated, and the efficiency runs about 3 or 4 per cent.

A one-half horse-power compressor of 60% efficiency provides about 200 watts in gas energy, so with 4% efficiency, we can obtain about 8 watts from the combination up to 25 kc or so, and less at higher frequencies.

Since this represents about the best available experience in the art, and is pretty poor in many ways, in what other directions can we go?

Explosions are certainly promising from the viewpoint of power. The Germans used a cavity in the hull of a ship which they pumped full of gas and set off with a spark plug. This was of course pulse work at a rate of one per second, and the average power would be unusably small.

We cannot consider slow combustion, as in an engine cylinder, because the propagation rate is only 70 feet/sec. A true explosion must be used, and the problem of setting up 25,000 such explosions per second is quite difficult. The valve would have to operate at extremely high temperatures, and must shut completely to prevent exploding the gas on the other side of the valve.

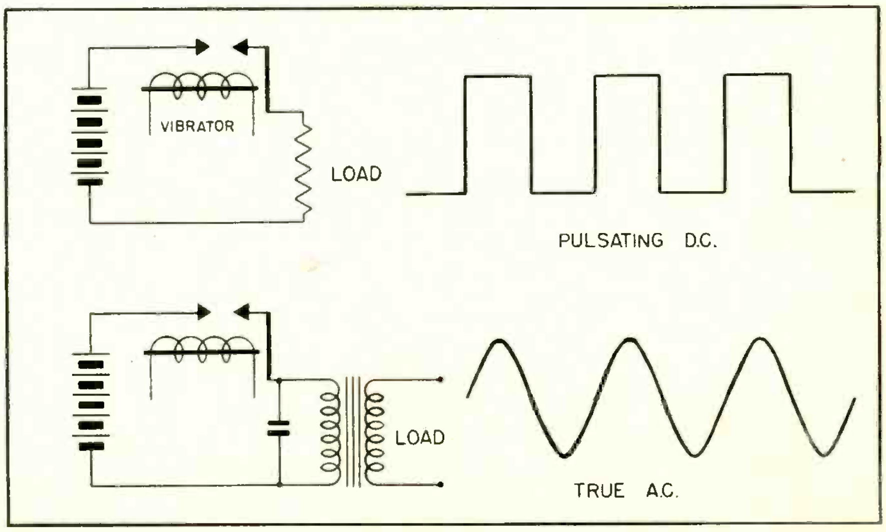



Fig. 2. Siren operation is similar to that of a vibrator. When an elastic load is used, the wave form is greatly improved.

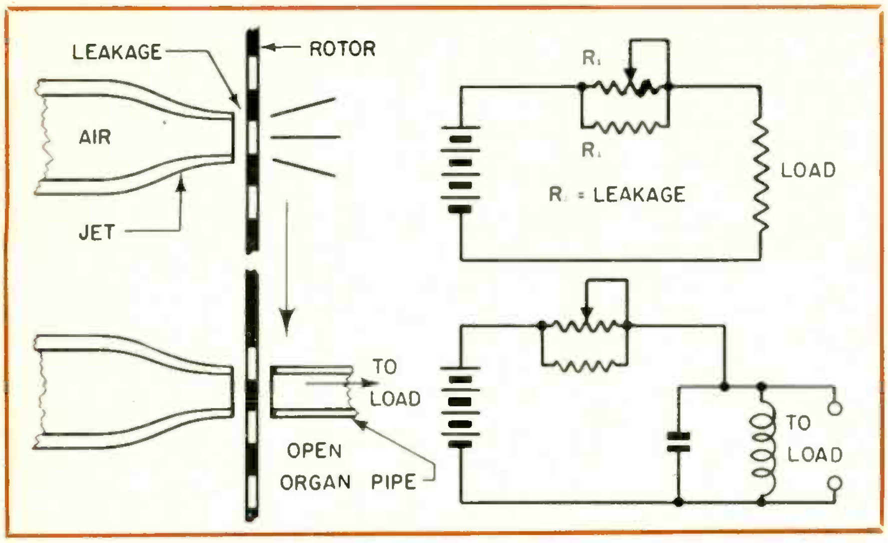



Fig. 3. Equivalent electrical circuit of turbine (above) and (below) electrical equivalent of effect obtained when a tuned organ pipe is used as a coupling medium.

The energy is there, though. A mol of acetylene and oxygen gives 120 million watts, and the products are water and carbon dioxide, both easy to handle.

A stream of magnetic gas can be deflected by an audio or r-f field of considerable intensity and might give a kilowatt or so. Also a compressed air loudspeaker unit might be made to oscillate through a small range by a magnetostriction rod.

When we consider power production from gas we think of the steam turbine where 50,000 h.p. is common, or, in reverse, 5,000 h.p. air pumps for blowing steel. In the design of either unit, when high peripheral velocities are used, turbulence becomes a factor to be designed around. On investigation there are apparently two types of turbulence, the random and the continuous wave. Blades can be stripped, and the efficiency can decrease by 40% or more when turbulence takes place. Here is certainly real power.

### Sirens

A similar rotary device is the siren, although never made in a thousand-kw size as yet. It is a gas chopper, similar to a vibrator in a vibrator power supply. It does not generate a.c. directly; it merely chops d.c. and gives us pulsating direct current. Let us look at *Fig. 2*: if the vibrator works into a resistive load we obtain a practically square wave form. Since such a wave form is undesirable for most apparatus, we work the vibrator into an elastic load, a tank circuit as shown, which gives us a fair sine wave. Note that theoretically the efficiency is perfect, since the vibrator neither adds nor absorbs electrical power (neglecting sparking and driving power). It is a very simple converter. Note also it is all open or all closed, having theoretically no intermediate condition.

As shown in *Fig. 3*, the siren has some points in common with the steam turbine. Unmodulated gas is always escaping through the leakage provided by the necessary clearances, and acts as an unchanging resistance, *R2*, which might go through the load or be by-passed to ground, depending on the construction. It is fundamentally impossible to open the orifice instantly, so we obtain a varying resistance, *R1*, in series with the load. The curve of this resistance change can be anything within wide limits, as we can have various shaped orifices. A round orifice and round holes will give a shape somewhat similar to that shown.

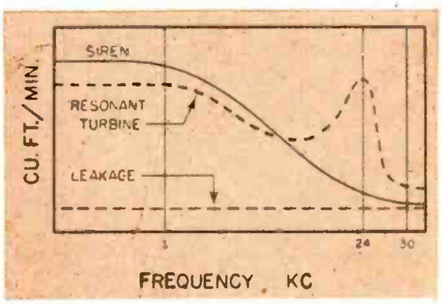



Fig. 4. Effect of varying frequency of sound generators.

The lower part of *Fig. 3* shows such a combination working into a tuned open organ pipe, corresponding to the resonant tank of *Fig. 2*. This makes the wave form somewhat more pure, and the outer end of the pipe gives us a source of known characteristics to feed energy into our load.

When we begin to apply really high power to the arrangement, a rather curious factor enters our organ-pipe design which has no equivalent in electrical circuits. We must pass so much air through the organ pipe that the windage velocity begins to be a substantial fraction of the velocity of sound itself. If we actually exceeded 1100 ft/sec air flow in the pipe, there could be no sonic wave in it, as the wave would be blown out of the pipe entirely. We have had average velocities of substantially half sonic (550 ft/sec), however, so a wave going upstream would have a velocity of 1100 minus 550, or 550 ft/sec, and downstream 1100 plus 550, or 1650 ft/sec. So a fixed length pipe would be, say, one-quarter wave one way and three-quarters wave the other, since there is a relation of 3 to 1 in the wavelengths, depending on direction of travel. It is not as simple as this, as the wind blowing through is intermittent but the effect is most important in the design.

When we set up a siren capable of reaching, say, 30 kc, and vary the r.p.m. and consequently the frequency, we observe a curve as in *Fig. 4*. We simply cannot force much air through at high frequencies.

Then we realize that at 400 cycles we have 2500 microseconds per cycle and the time required to accelerate the gas is a negligible portion of this enormous time interval. At 24 kc, however, we want to accelerate the gas to sonic velocity in about a quarter wave, which is ten microseconds. We have insufficient pressure to give us this enormous acceleration of thousands of miles/sec$^2$. So we step up the pressure and are disappointed again, because as we increase the pressure of the gas we also increase its density to the same degree, so we can never obtain sonic velocity, where our hope of real power lies.

But a sonic wave consists of a pressure portion and a rarefaction portion. If we think in terms of kw per square centimeter, our sonic positive pressure is going to run quite high—perhaps a hundred pounds/in$^2$. The average pressure in the gas will remain unchanged up to the point where we modulate it 100%. So if we can add great sonic pressure to the static pressure at the instant our orifice opens we have hopes of much increased acceleration.

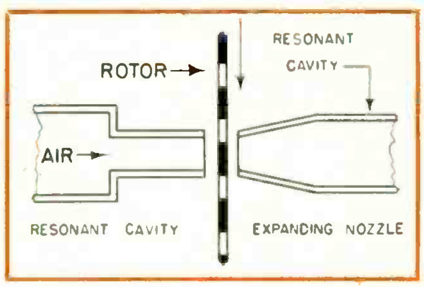



Fig. 5. Method of increasing acceleration.

So we move our organ pipe over to the input side, as in *Fig. 5*. Now we have a structure quite simple in form but quite a bit too difficult to calculate at the present state of the art. So we

[*Continued on page* 45]

# MUSICAL ACOUSTICS

**BENJAMIN F. TILLSON***

**PART II**

**This is the second of a series of articles on music theory, written especially for sound engineers.**

Music is a purely human, pleasure-evoking, artistic emotion of auditory perception. Although Nature may offer sounds which can inspire composers, there is lacking the sustained combination of rhythm, melody, and harmony so necessary in music.

Music existed centuries before the scale was conceived, although it probably lacked the harmony we now consider necessary. Savages possessed no scale and wished none; for much music of other peoples cannot be presented by our European scale. Medieval Europeans sought for a scale in vain. Even a Chinese octave is a little out of tune, according to their orthodox theory.

Thus a scale has become an abstract formulation from experiments and compromises. Music appreciation and ear-training is a matter of experience, study, and environment. With the growth of civilization it has progressed in refinement and complexity. Like other cultural arts, enjoyment is heightened by increased study, understanding, and association. Like them, an unorthodox or perverted taste may be developed in the manner described by the verses of Alexander Pope:

"Vice is a monster of so frightful mien
As, to be hated, needs but to be seen;
But, seen too oft, familiar with her face,
We first endure, then pity, then embrace."

The scales used in China, Japan, Java, and the Pacific Islands are pentatonic (five tones to an octave); and most of the rest are heptatonic (seven-toned), as in India, Arabia, probably Egypt, certainly ancient Greece, and in modern Europe; although hexatonic (six-toned) scales also exist. The ancient Persians discovered that an ideal scale for all keys was impossible, but they had an excellent seventeen-tone scale that gave true intervals for the Fifth, Fourth, Major Third, Major Sixth, and Minor Seventh. Their modern scale of 24 equal quarter-tones does not give those intervals as truly.

Our present scale is based upon the ancient Greek scale, made up of two tetra-chords (four-tone chords with two whole-tone and one so-called half-tone intervals) separated by a whole-tone interval, with the final tone an octave higher than the initial tone. But the whole-tone intervals were not and are not of equal size, nor the half-tone exactly half of either of the whole tones.

Didymus, who lived 63 BC to 10 AD, is the first to mention such a seven-tone scale in which the frequency ratios of successive tones in an octave were increased by the following fractions:—⅛, 1/9, 1/15, ⅛, ⅛, 1/9, 1/15. Claudius Ptolemy, c. 73—c. 151 AD, formulated his new scale, but Pythagoras, 582-500 BC, had discovered the relations by experiments on a vibrating string (monochord) tuned to E. Pythagoras found that the tone from half its length was an octave higher, that two-thirds its length gave B the fifth above E, and that three-quarters of its length gave A the fourth above E. Because of their simple numerical relation he called the interval A to B a whole tone and, noting that the intervals of E to A and of B to the octave of E each represented two of such whole tones plus a fraction, he called such fraction a semitone. Pythagoras did not prescribe the arrangement of such tones as in our major scale; but the Greeks developed nine such arrangements of intervals in octaves and called them modes. By the 16th century the Church was using twelve "modes", as they were called up to the time of J. S. Bach (1685-1750).

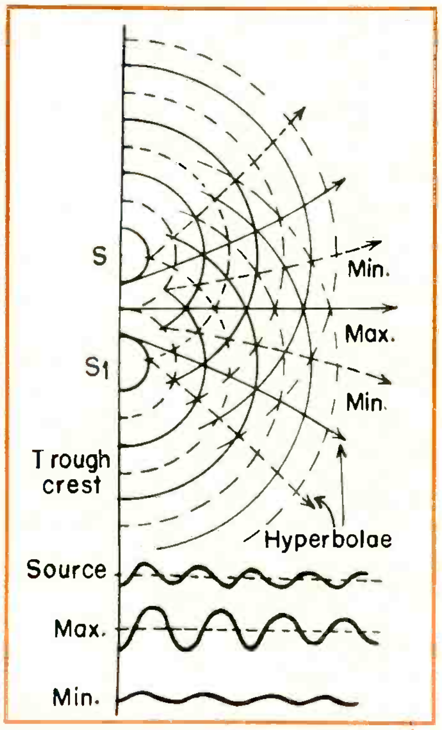



Fig. 3. S1 and S2 are sound sources, separated two wavelengths. The sound waves combine vectorially as shown. Dotted lines represent troughs; solid lines, crests, of waves.

Claudius Ptolemy's improvement of Didymus' scale was by transposing the fifth and sixth intervals so as to give the series:—⅛, 1/9, 1/15, ⅛, 1/9, ⅛, 1/15. This made all Fifths, Fourths, Thirds, and Sixths pure, except the Fourth above F and the Fifth above B, which are tritones as in our even-tempered scale. It resulted in a series of six ascending pure thirds, alternately major and minor, beginning on F.

Only one other seven-tone scale of "just intonation", of three major tones (ratio 9:8), two minor tones (ratio 10:9), and two semi-tones (16:15), is possible, although it has not been adopted. In it the sequence would be:—1/9, ⅛, 1/15, ⅛, 1/9, ⅛, 1/15. Further discussion of this new series proposed by John Redfield is found in his book "Music".

Both of the above seven-tone scales show the following simple arithmetical frequency relations between the consonant notes of the larger intervals:—E:C::5:4; G:C::3:2; G; E::6:5; C′:G::4:3; C′:C::2:1

We further note that when tones have a frequency ratio of any power of 2 (therefore are octaves) they may replace each other in a chord and leave it concordant and pleasing; also that any three notes are concordant in a chord when their frequency ratios are 4 to 5 to 6, as C–E–G, which forms a major chord when the octave C′ is added to make C–E–G–C′. And there is another frequency ratio of three notes which, while not exactly concordant, are only so slightly dissonant as to produce a somewhat pleasing auditory sensation. Their frequency ratios are the fairly simple ones of 10 to 12 to 15, represented by the series A–C–E, or C–E flat–G, which form minor chords when the respective octaves A′ or C′ are appended.

### Modulation and Temperament

When the key is changed within a musical composition, or when the sequence of intervals is changed from those of a major to a minor mode, such a change is called a "modulation."

*Consulting Engineer, Montclair, N. J.

An "intonation" is any precisely specified system of tuning the intervals in the various scales. A "temperament" is an intonation which has been deliberately qualified with the sacrifice of true intervals to compromised ones for the sake of gaining simplicity in the writing and reading of music and in the technique of performance on instruments, particularly such instruments as have a keyboard, fretted finger-board, or fixed and pedal control (as a harp or xylophone). An ideal intonation would exhibit:—1) pure concords (exact intervals with simple ratios of frequencies); 2) free modulation (which would permit harmonious changes to any other key); 3) practical convenience (which is generally interpreted to mean limiting the number of notes in an octave so as to simplify tuning, and the writing of music for an instrument and its playing).

But the keyboard instruments, primarily the piano and organ, are chiefly responsible for the twelve-tone octave, since there is no limit to the intermediate pitch variations of tones on all wind instruments by a controlled embouchure and a critical ear, as well as with all unfretted finger-board stringed instruments. Such a possibility for variations is not commonly realized for other than the violin family and the slide-trombone.

Since some of the intervals are repeating decimals it is impossible to represent the steps exactly. But if the octave were divided into 301 steps (one thousand times the approximate logarithm of 2) it would be found that all the notes required for the fifteen major key signatures would be available for a close approximation to "just intonation" if the octaves were divided into twenty-five tones.

However, for perfectly free modulation an unlimited number of steps in a scale is required. But Bosanquet showed that a 53-note octave could be modulated in 53 keys with all of the intervals practically true (three notes flat by five-thousandths of a tone); while equal temperament with 12 notes per octave will modulate in 29 keys, but offers very sharp intervals for the major third, the sixth, and the seventh.

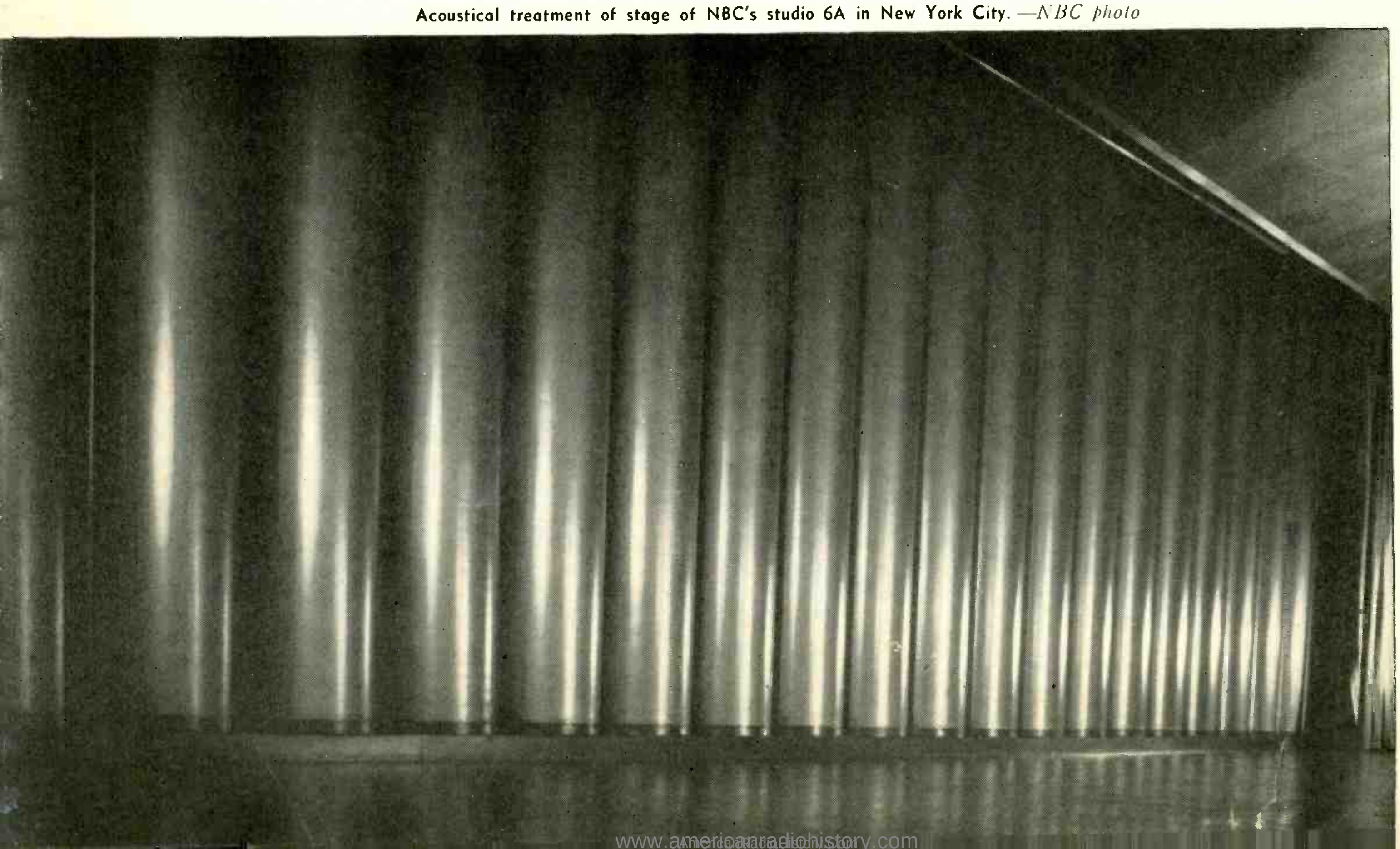
### Equally-Tempered Tuning

In equally-tempered (or, as a scientific wit dubbed it, "equally-tampered") tuning the intervals of the fifth and the fourth in the octave above middle C are each made to give about one "beat" per second. This is so slight an error as to be hardly noticeable; but the thirds and sixths are badly out of tune. The major third, D to F sharp, gives nearly 12 beats per second, which are rather strong and distinct and become still harsher if the interval is extended to a tenth or a seventeenth. The major sixth of A below the treble staff up to F sharp gives about 10 beats per second which are so violent that this interval in its tempered form barely escapes being regarded as a dissonance. And the Difference-tones (which will be described later as the beat-frequency resulting in a new tone from other frequencies) from such "tempered" chords are also thrown very much out of tune; and even when too far apart for their beats to be counted still produce a disagreeable effect.

However, equal temperament meets the following requirement of modulation. If one proceeds upward on a keyboard by intervals of the perfect fifth he will arrive in twelve such steps at a note seven octaves above the note from which he started. If G flat was the starting note he will arrive at F sharp, which is identical to G flat on the keyboard, and therefore called "enharmonic" with it. This is accomplished because each fifth is tuned somewhat too flat for consonance. If the fifths were tuned at perfect consonance the final F sharp would be sharper than the first note G flat by a small interval called Pythagorean Comma, which is 23.46 per cent of a semi-tone. Hence each of the twelve fifths should be flatted about two percent of a semi-tone. Such accuracy proves impractical so variations occur in tuning and the errors in some fifths are greater than in others. Accordingly, the dissonance in tuned keyboard instruments is greater in some intervals than that prescribed for equal-temperament, which should divide the octave equally into twelve semi-tones (whose successive ratios are the twelfth root of 2), six tones, or three major thirds.

The theory of "equal-temperament" was proposed by Mersennes in 1636, was greatly promoted by J. S. Bach (1685-1750), and did not appear in England until 1846. This system of temperatment superseded that of the "mean-tone" system which is credited to the Spaniard, Salinas (1513-90), although some attribute it to Guido d'Arezzo (995-1050). In the "mean-tone" system, the major third (say D to F sharp) was made perfect and the

**Acoustical treatment of stage of NBC's studio 6A in New York City.** —*NBC photo*



four intermediate fifths were made equally flat by trial to arrive at the F sharp two octaves higher. One difficulty of such a system was that it permitted modulation in only six major scales and three minor scales. Other systems of temperament have been proposed by Bosanquet, Woolhouse, and Huyghens; but they have had no extensive use.

### Scale Formation

Lest one consider a scale of "natural" or "just" intonation entirely arbitrary based upon a fancy for simple arithmetical ratios of frequencies, it would be well to note how it is derived from the natural vibrations of strings or air-columns.

With but few exceptions, like a very softly blown flute, pure tones of simple frequency ratios are rarely produced by the classical musical instruments. Rather are a mixture of tones produced with the fundamental and the higher frequencies of its harmonics (over-tones). The variation in the frequency and intensity of such partials can be proven by harmonic analysis and synthesis to be responsible for the different characteristic tone colors (or timbre) which distinguish the various instruments or different registers in any one instrument. The frequencies of these partials bear a whole number relationship to the frequency of the fundamental. Successive multiples of 2, therefore all the powers of 2, represent successive octaves.

Harmonics on a vibrating violin string are produced by lightly touching the string at proper points where the damping of its amplitude produce stilled nodes of a more frequent series of vibrations throughout the entire free length of the string, which thereby vibrates in more segments. Categorically, by touching the string where it would be pressed down to bow its octave, fifth, fourth, major third, or minor third, it will vibrate respectively in two, three, four, five, or six sections, and will produce, respectively:—the prime and octave; the octave and twelfth; the twelfth and fifteenth; and the seventeenth and nineteenth partials.

### Natural Harmonic Diatonic Scales

The Fifth below C' is F', which is called the sub-dominant of the Key of C'. But its relationship is even more important because the diatonic scale of C is exactly represented by the sub-octaves of the harmonics (or partials) of F, the sub-octave of F', as follows:—

| | | | | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|---|---|
| **Partials** | 1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 | 9 | 10 | 12 | 15 |
| **Tone** | F | F' | C' | F'' | A'' | C'' | $E^{b}$* | F''' | G''' | A''' | C''' | E''' |
| **Frequency** | 44 | 88 | 132 | 176 | 220 | 264 | 308 | 352 | 396 | 440 | 528 | 660 |
| **Partials** | 16 | 18 | 20 | 24 | 27 | 30 | 32 | 36 | 40 | 45 | | |
| **Tone** | F'''' | G'''' | A'''' | C'''' | D'''' | E'''' | $F^{v}$ | $G^{v}$ | $A^{v}$ | $B^{v}$ | | |
| **Frequency** | 704 | 792 | 880 | 1056 | 1188 | 1320 | 1408 | 1584 | 1760 | 1980 | | |

(Note: scale $E^b$ is 309.03 or 312.89)

All major harmony is multiples of Partials 1, 2, 3, and 5; all minor harmony is of Partials 3, 5, and 15; and all harmony of the dominant seventh is of Partials 1, 3, 5, and 7.

With multiple division by 2 we can transform the higher notes to their frequencies in a common octave of C equal 264 and 528 to find:

| **Tone** | C | | D | | E | | F | | G | | A | | B | | C' |
|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|
| **Frequency** | 264 | | 297 | | 330 | | 352 | | 396 | | 440 | | 495 | | 528 |
| **Interval Ratio** | | 9/8 | | 10/9 | | 16/15 | | 9/8 | | 10/9 | | 9/8 | | 16/15 | |
| **Solfege Name** | Do | | Re | | Mi | | Fa | | Sol | | La | | Si | | Do |

We then see that the ratios of the frequencies of the various intervals are exactly those of the diatonic scale of "just intonation".

If we use these interval ratios to build a series of scales for the sharp keys in the order of the keynotes C, D, A, E, B, F sharp, and C sharp using always for the frequency of the keynote the one found in the next preceding scale, and similarly for the flat keys in the order C, F, $B^b$, $E^b$, $A^b$, $D^b$, $G^b$, and $C^b$, we find that the frequencies of the flats are lower than those of the sharps of the diatonic notes next lower in the scale; and we also find 35 variations of frequencies in an octave, as follows:

**Natural Harmonic Chromatic Scale Frequencies**

| C | B sharp | $D^b$ | C sharp | D | $E^b$ | D sharp | $F^b$ | E |
|---|---|---|---|---|---|---|---|---|
| 264.0 | 264.30 | 278.12 | 281.92 | 297.0 | 309.03 | 313.24 | 329.63 | 330.0 |
| 260.74 | | | 278.44 | 293.33 | 312.89 | 317.16 | | 334.12 |
| 261.63* | | 277.18* | | 293.66* | 311.13* | | 329.63* | |

| F | E sharp | $G^b$ | F sharp | G | $A^b$ | G sharp | A | $B^b$ | A sharp |
|---|---|---|---|---|---|---|---|---|---|
| 352.0 | 352.40 | 370.83 | 371.25 | 396.0 | 417.19 | 417.66 | 440.0 | 469.33 | 469.86 |
| 347.65 | | | 375.89 | 391.11 | 412.03 | 422.88 | 445.5 | 463.54 | |
| 349.23* | | 369.99* | | 392.0* | 415.30* | | 440.0* | 466.16* | |

| $C^b$ | B | C' |
|---|---|---|
| 494.44 | 495.0 | 528.0 |
| | 501.19 | 521.48 |
| 493.88* | | 523.26* |

Note: Those frequencies followed by * are the values for a tempered scale of the same octave as the natural scale of just intonation.

A divergence of frequency of 11 per cent of a tempered scale half-tone is large enough to be clearly discerned melodically by an acute musical ear, and is quite objectionable when compared notes are sounded together. The possible divergence of equal temperament from just intonation, expressed in percent of adjacent half-tone, is as follows:—

From maximum:

| B# | C# | D | D# | E | E# | F# | G | G# | A | A# | B |
|---|---|---|---|---|---|---|---|---|---|---|---|
| —17 | —29 | —19 | —33 | —23 | —15 | —27 | —17 | —31 | —21 | —13 | —25 |

From minimum:

| C | $D^b$ | D | $E^b$ | $F^b$ | F | $G^b$ | G | $A^b$ | A | $B^b$ | $C^b$ |
|---|---|---|---|---|---|---|---|---|---|---|---|
| +6 | —6 | +2 | +11 | 0 | +8 | —4 | +4 | +13 | 0 | +10 | —2 |

It is apparent that the equally-tempered scale departs less from and is more consonant with the diatonic values of just intonation in the flat signatures than in the sharp signatures, where their tendency is to be flat (too low).

Since the ratio of frequencies for each of the twelve half-tone steps in an equally-tempered scale is 1.059463, the key of any phonograph reproduction will be raised a half-tone if a recording at 33 rpm is played at 34.96 rpm, or a 78 rpm recording is played at 83.34 rpm.

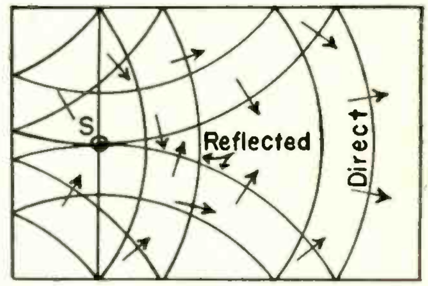
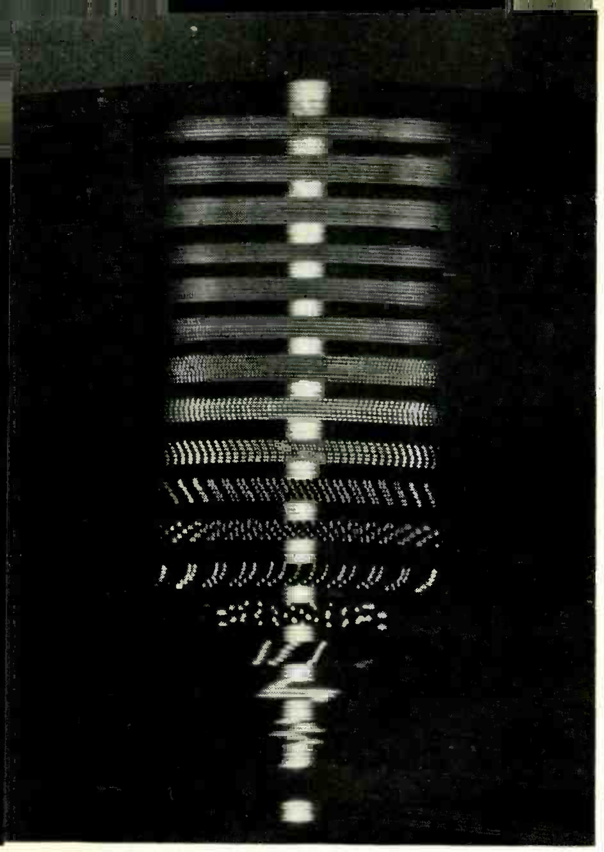



Fig. 4. Sound wave interference patterns in an auditorium 60 feet long and 40 feet wide, when the sound originates at point S, 12.5 feet from an end wall.

### Beats and Subjective Tones

Oscillatory waves may be combined algebraically to give resultants equal to the sums or differences of the vectors of the amplitudes in the same respective phase. In *Fig. 3* are shown two sources of sound located two wavelengths apart. The solid-line circles represent the crests of the waves, and the dotted-line

[*Continued on page 42*]



# Applications of the FM Calibrator

**Uses of the FM calibrator described in the May and June issues are described in detail, with typical test results.**

Fig. 7. Optical pattern of calibrated frequency record made with the aid of the FM Calibrator. *Courtesy Journal SMPE*

**RALPH A. SCHLEGEL***

ANY ONE WHO HAS LABORED through the mechanics of measuring the response of recording heads by the time-honored method of reflected light pattern comparison, or by recording a series of frequencies, which are measured by reproduction through a previously calibrated reproducing system of perhaps dubious accuracy, will appreciate the advantages offered by the FM calibrator. This device permits making various observations of a recording head's behavior during actual recording conditions.

Briefly, the calibrator may be used to investigate:

1. Changes in frequency response due to cutter load variations.
2. Changes in frequency response due to cutter styli lengths.
3. Effects of various disc coatings upon frequency response.
4. Distortion in the recording head.
5. Input-output linearity.
6. Effect of room temperature upon frequency response.
7. Comparison of electrical output of reproducers to the mechanical vibration of the reproducer stylus.

In addition, the calibrator may be used for:

1. Preparing calibrated test records.
2. Repairing and adjustment of recording heads.
3. Feedback to the recording amplifier.†

### Frequency Response Curves

Frequency response curves obtained with calibrator are shown for four types of recording heads in general use today. *Fig. 1* illustrates the effect of groove velocity upon the frequency response of a cutter used by a great many professional and home recorders. The effect of recording speed, or groove velocity, is particularly noticeable in the high frequency region around 6000 cycles. *Fig. 2* shows the variation at this frequency for different recording diameters while recording at 33 1/3 rpm.

Curve *"B"* of *Fig. 1* shows the low-frequency response obtained with this head when the compensating network is removed from the output of the recording amplifier in order to increase the low frequency response for 78 rpm recordings. The recording head's impedance decreases rapidly with decreasing frequency, presenting what amounts to a short circuit to the amplifier's output terminals when the compensator is removed. *Fig. 3* shows oscilloscope patterns of stylus motion for 200, 150, 100 and 50-cycle sine waves. The increasing distortion with decreasing frequency is very apparent when this method is used to obtain increased bass response. It would be better to insert an equalizer in the recording amplifier's input circuit and thus permit the amplifier to operate with a proper impedance termination. The oscilloscope traces were obtained while using a fifty-watt recording amplifier to drive the recording head. One may well wonder what the pattern might have shown, had a ten watt amplifier been used.

*Fig. 4* shows the frequency-response characteristic obtained for the new Presto 1D recording head. The 6000-cycle amplitude varies somewhat with groove velocity, approximately 2.5 db change was observed at 33 1/3 rpm for

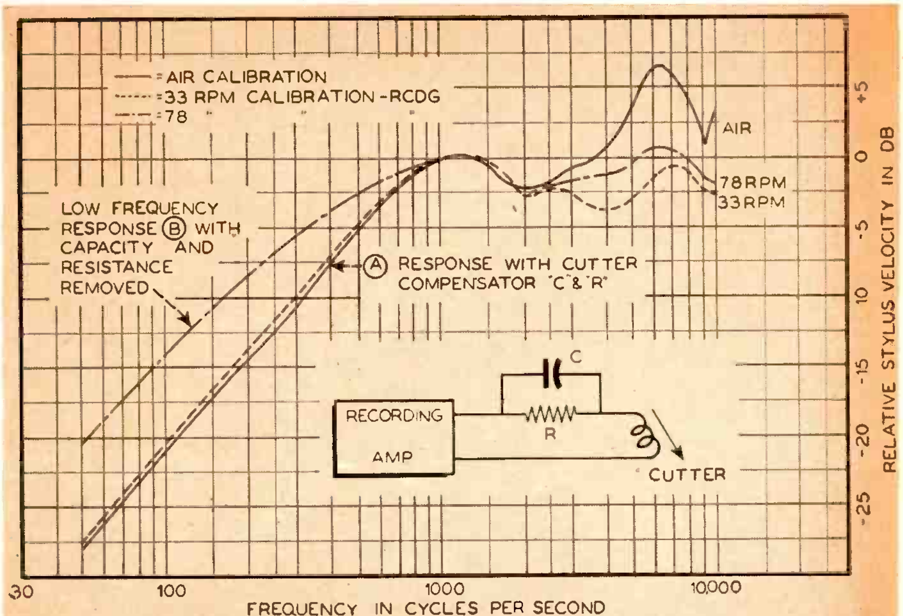



Fig. 1. Response characteristic of recording head showing variations due to changing groove velocity.

*WOR Recording Studios, New York City.

†US Patent #2,400,953, H. E. Roys

a recording diameter change from 15.5″ to 6.5″. A similar change was noted while going from 12″ to 3¾″ diameter at 78 rpm. The response below 400 cycles is the same in air as for recording.

*Fig. 5* shows the response characteristic of the Fairchild recording head, which exhibits a peak at 8000 cycles when the stylus vibrates freely in air. This peak drops to a negligible value when recording. The low-frequency response of this cutter increases slightly when recording at 33 1/3 rpm; at 78 rpm, the air and recording calibrations are identical below 350 cycles.

*Fig. 6* represents the frequency response characteristic of the new R.C.A. recording head with thermostatic temperature control of the viscaloid damping material. The air calibration of this head is within 1 db of the recording calibration. Changing groove velocity affects the 1000-cycle region on this head, the change in response at 1000 cycles is approximately 2.5 db from outside to inside recording diameter at 33 1/3 rpm. Frequency response measurements were made at room temperatures ranging from 63 to 80 degrees over a period of a week and all measurements checked within a few tenths of a db of the curve shown during these tests.

### Varying Styli Lengths

Some measurements have been made on the effect of different styli lengths. Economy-minded recorders have used styli that had been resharpened so often that only a tiny portion of the jewel was visible in the metal shank. Tests were made with several of these shortened jewels and an appreciable increase in high frequency peaks was observed. One jewel showed a peak in the 8000-cycle region in excess of 10 db while other short jewels showed similar effects.

The calibrator lends itself admirably to the study of the effect on the recording head of various disc coatings. Measurements have been made toward this end and indicate some variation in response starting at 4000 cycles and being most noticeable in the high-frequency region around 12,000 cycles, where a 3.5 db increase in recording head response was noted on an experimental soft coating to be used for master discs.

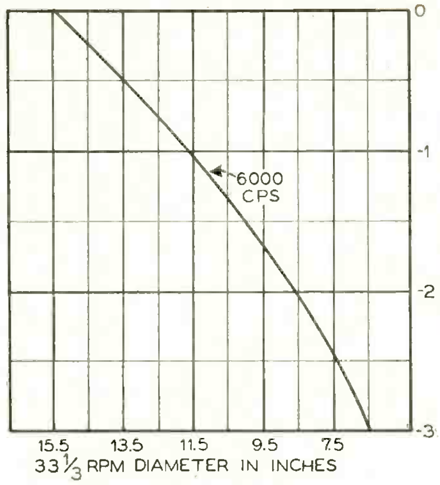



**Fig. 2. Effect of recording diameter upon cutter head resonant peak.**

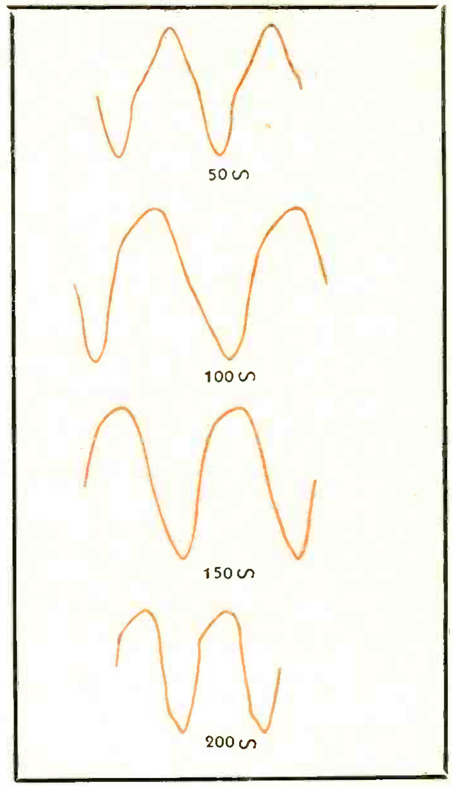



**Fig. 3. Oscilloscope patterns of stylus motion when compensating network has been removed from amplifier output circuit.**

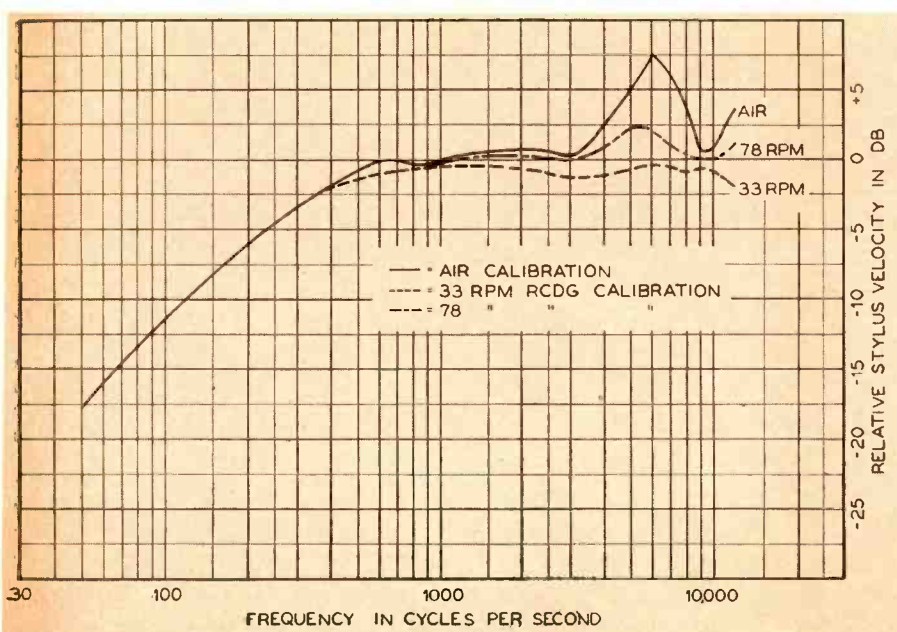



**Fig. 4. Response curve of Presto 1D recording head.**

In the past, distortion in recording systems was difficult to analyze as it was impossible to segregate distortion of the cutter head, cold flow of the recording media or non-linearity of the reproducing system. When using the calibrator for distortion measurements it has been found advisable to use a plate spacing of 0.015″ when checking on the constant amplitude portion of the recording head's characteristic. The maximum stylus vibration should not exceed 3 mils in either direction at the lower frequencies while smaller plate spacing may be used when measuring distortion over the constant velocity portion of the cutter characteristic. An average of slightly less than one per cent harmonic distortion will be found in the mid-frequency range of most recording heads; intermodulation distortion will usually be in the order of 3 to 4.5%. On some types of recording heads, it is possible to measure the intermodulation distortion in air. Some cutters will show as high as 25% intermodulation under this condition, dropping to 10% when recording. A condition of this sort usually indicates insufficient damping or misalignment of the vibratory system of the head. Input-output linearity is easily checked with the calibrator by increasing the input to the recording head in 1 db steps and plotting against the output as read on the calibrator output meter. A good recording head should show a linear input-output characteristic for at least 8 or 9 db beyond its normal operating point.

The FM calibrator has been used to

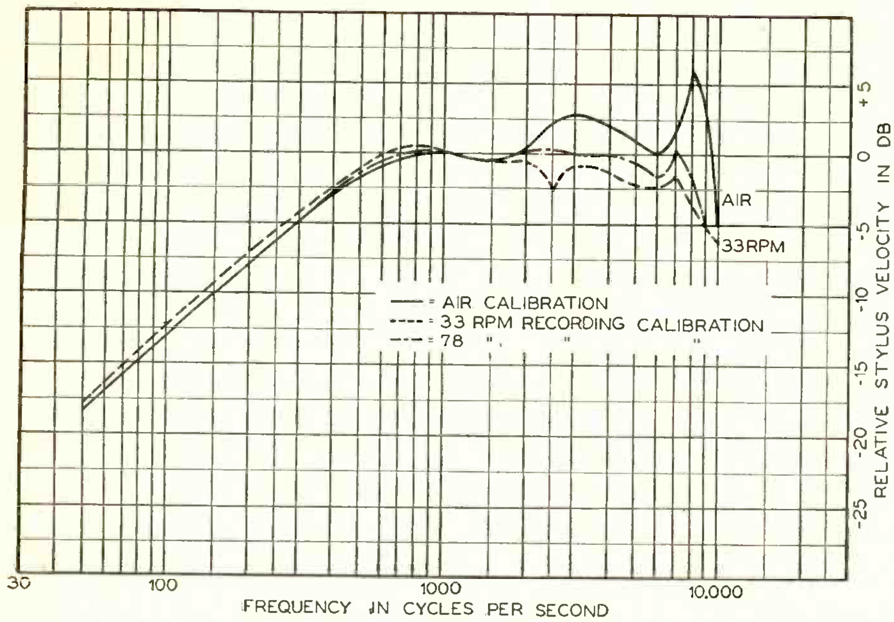



Fig. 5. Response curve of Fairchild cutter.

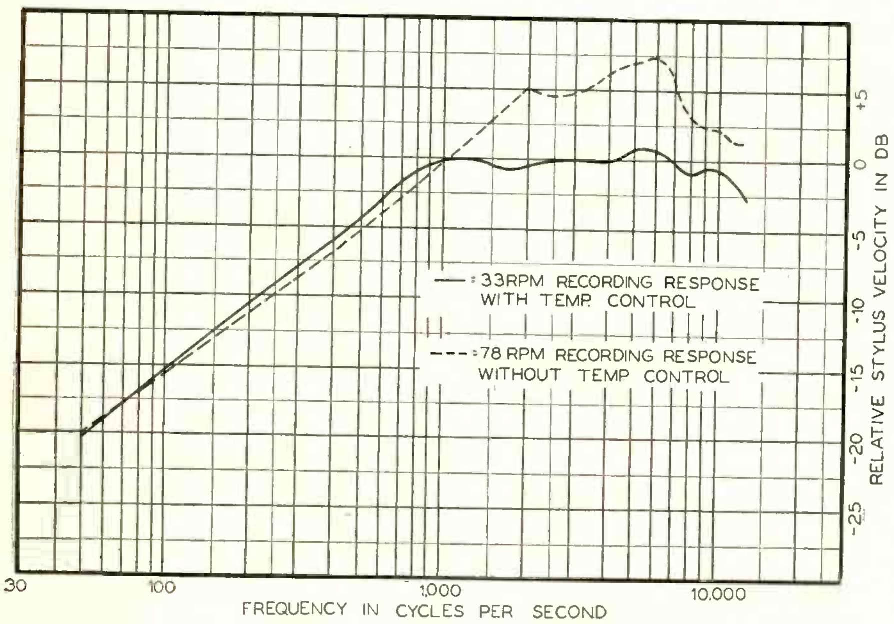



Fig. 6. Frequency response characteristic of RCA MI 11850C recorder, showing curve obtained with and without temperature control of viscoloid damper.

compare the electrical output against stylus motion of a reproducer. To accomplish this, small insulated wires were used in place of the push-pull condenser plate assembly used for cutter head analysis and were held in position with small pieces of scotch tape.

#### Calibrated Test Records

The calibrator makes it possible to make frequency records for testing and adjusting reproducing systems. To make a calibrated test record, it will be necessary to calibrate the recording head. With this information at hand, apply such corrections to the input of the recording head as may be required for the test record. The calibrator may then be used as a final check against the cutter corrections. *Fig. 7* shows a frequency record made in this fashion, the constant velocity portion is flat within a few tenths of a db.

To those who, by choice or circumstances, must repair and adjust their own recording heads, the calibrator will be a welcome addition to their test equipment. Used in conjunction with an audio sweep frequency oscillator, such as was used to make the "Clarkstan" sweep frequency record, it would be possible to view the entire recording characteristic on the screen of a cathode ray oscilloscope. With such a set-up, one may watch the response curve change as adjustments are made to the mechanical system of the recording head.

A patent† has been issued to H. E. Roys covering the use of the calibrator as a feedback device. Considerable difficulty would be experienced if one did not take into consideration the phase shift prevalent in recording heads.

Recording heads using feedback provide an excellent means of measuring stylus motion during recording. It is claimed that the feedback recorder is flat on a velocity basis and it is possible to measure stylus motion on this type of recorder by metering the feedback loop. The writer hopes to have the opportunity of comparing FM calibrator readings with feedback loop readings, and have the test record so made, optically calibrated. With this information correlated, it is possible to arrive at a standard method of measuring recording head performance.

## Iron In Tube Design

Pure degassed iron can effectively replace molybdenum and nickel in the construction of electrodes, getters, and other metallic parts of electron tubes, according to a Japanese research document included in a report now on sale by the Office of Technical Services, Department of Commerce.

The report, prepared by Major Wilhelm Jorgensen for the U. S. Naval Technical Mission to Japan, consists of five studies by leading Japanese experts in electronics.

Shortages of nickel and molydenum during the war compelled Japanese engineers to investigate the use of iron for vacuum tube parts. They discovered that iron, when freed of occluded gases, possessed the durability and the electron emission characteristics comparable to those of molybdenum and nickel.

The raw material for the iron used in electron tubes was iron sand. The sand was reduced in an electric arc furnace and the iron metal drawn off. The metal was then melted in a high frequency electric induction furnace and deoxidized with silicon and aluminum. As a result of this treatment the iron contained less than 0.02 percent of oxygen—the most harmful of the occluded gases—and therefore became a suitable construction material for small electron receiving tubes.

However, large power tubes required iron with an even lower oxygen content. To obtain such purity, the iron after deoxidation with silicon and aluminum was heated in a stream of hydrogen at 1,100 degrees Centigrade for two hours. The final oxygen content was less than 0.004 per cent.

The report also describes high frequency magnetrons built by the Tokyo Shibaura Electric Company, the Osaka Imperial University, and the Japanese Ninth Military Technical Laboratory.

Orders for the report (PB-49839; *Japanese Electronic Tubes*; photostat, $3; microfilm, $1; 35 pages including illustrations) should be addressed to the Office of Technical Services, Department of Commerce, Washington 35, D. C., and should be accompanied by check or money order, payable to the Treasurer of the United States.

# INDUSTRIAL MUSIC

**SAMUEL C. MILBOURNE**

**Installations to provide music for patrons of stores and other public places form a rapidly expanding market for sound equipment.**

SINCE THE WAR, the field for industrial sound has expanded far beyond the microphone-amplifier-speaker prewar conception. It is the purpose of this article to present the evolution of postwar industrial music, and thus bring the reader up-to-date.

As is well known, the basis for all industrial sound systems is (1) an input device such as a microphone or record player, (2) an audio amplifier with the proper power output, frequency response and other electrical characteristics, and (3) an output device, or devices, such as cone speakers in baffles, or horns and exciter units. Many variations of industrial sound systems are too well known for inclusion here. These include paging and music systems expressly or generally designed for industrial use.

*53 Grand Rd., Stamford, Conn.*

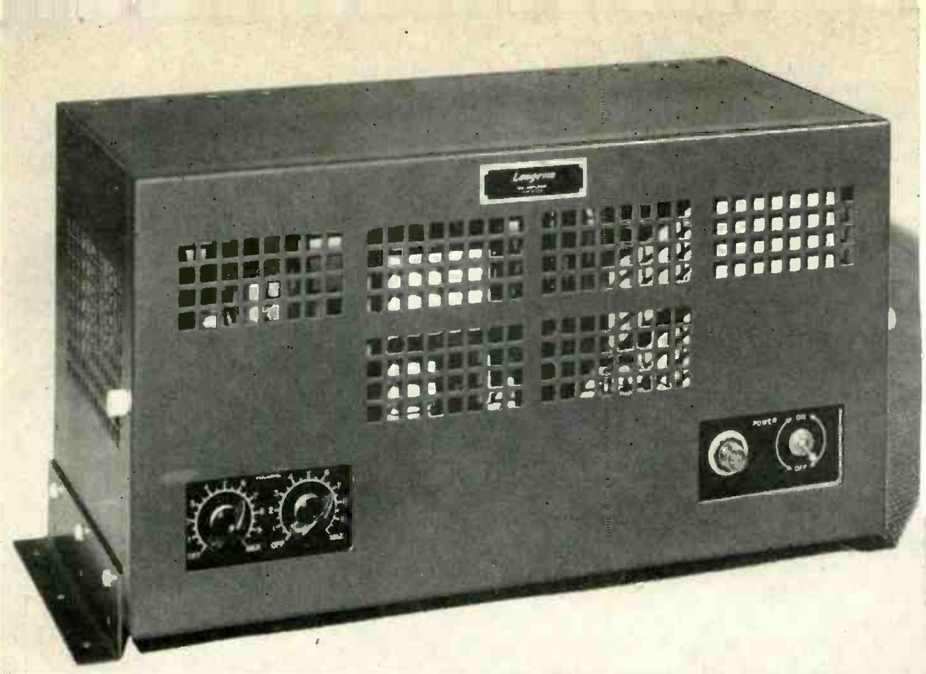
### Wired Music

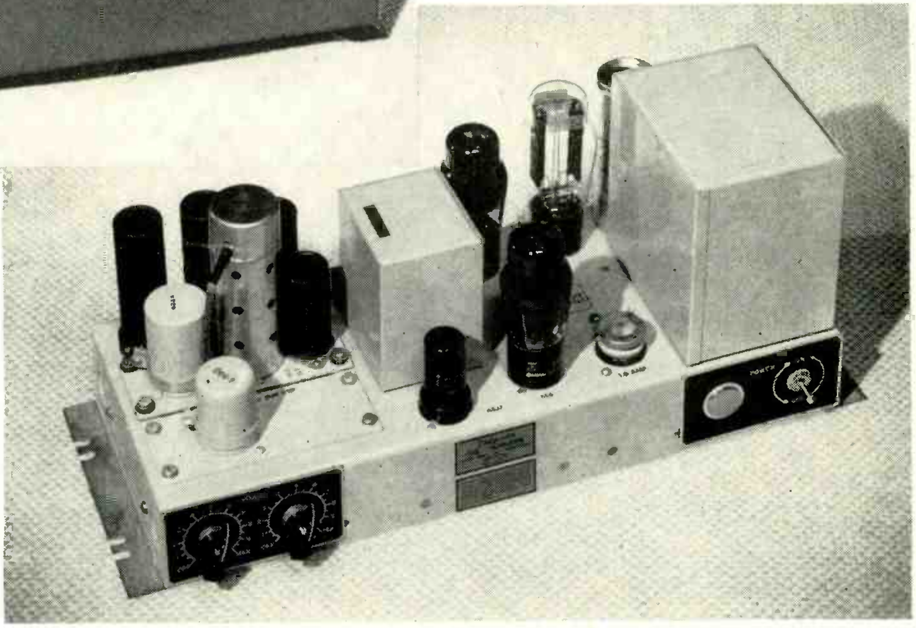
However, a variation of industrial sound service is the furnishing of amplified special musical programs to all types of public places such as industrial plants, professional men's offices, business establishments, stores, etc. This is, in general, called "Industrial Music" or, when supplied by telephone line, "Wired Music."

Industrial music is distributed in a number of ways. While, on first thought, the old, familiar "juke box" has little in common with modern industrial music systems, it can be said that the juke box with its gaudy colors and the kettle-drum response is the cornerstone of industrial music systems. The original boxes had coin slots only at the machine itself. To make it easier for the customer to spend his nickel, remote control of the machine, with coin slots and selection mechanism at each table, was developed. However, the number of selections is limited in such a set-up and it was felt that a location would be more profitable if the customers could have a wider choice.

Thus we come next to the "operator-controlled remote" type of juke. In this system, the customer requested his selection from an operator at some other place in the town, and the operator would play it for him. The basic system in this case consists of a coin box, speaker, amplifier and means for talking to the operator—these being at the

Installation of "hidden" speakers in ceiling of new Syl-May drug store, Stamford, Conn. For use with wired-music programs served by Telemusic of Stamford. Inset shows cutout in ceiling over which speaker is mounted.





Langevin Model 108-C amplifier, used for many industrial music installations, and (below) chassis of this amplifier.

location. At the central station there are turntables, operators for handling the requests, line amplifiers and switching circuits. The central station and the locations are connected by telephone lines which are leased from the telephone company. The disadvantage of this system is an economic one—it requires too many people to run, and generally costs the customer a dime to use, rather than the usual nickel. However, it shows definite evolutionary signs, inasmuch as it first brought into use the telephone line as a link between the record and the customer.

Of course, the obvious development from this point would be a similar system, but operated completely automatically by means of the dial telephone. In such a system, the customer might have a list of two hundred or more selections from which he could make his choice. Each song title would be followed on the list by its "telephone number". A nickel in the slot then would allow the customer to dial the number of the desired song, with the selection of the record accomplished automatically. Some work has been done on this type of system, but—if it can be worked out—the system will be a rather costly and bulky one.

### Metered Music

A slightly different approach to the previous system is being tried with fair success in at least one city. In the "Metered Music" system, a central station or studio "broadcasts" a continuous musical program—usually of the popular type—by telephone wire to all the locations. A nickel in the slot of a unit at any location brings the customer—at his table—a metered period (usually about six minutes) of music. The biggest drawback to such a system is the inability of the customer to pick his own selections. Another is that he may come in at any point in an individual recording and his time may be up in the middle of another record. However, it has the interesting angle that here is the link between the juke box and industrial music systems.

It is a short step between metered music and systems such as "Muzak" or "World Music." Instead of buying the music in small periods of time, the customer buys it by the month. Also, in place of the customers at the location being the music purchasers, the locations themselves buy the music. Instead of the nickel-in-the-slot mechanism, a steady flow of music—properly selected and balanced—is assured. The remainder of the system is essentially the same.

The generic term usually used to classify industrial music systems operating from a remote point and linking the record with the speaker by means of telephone lines is "wired music". As utilized by operators of the "monthly-basis" systems this is handled in the following way: The operator has a bank of automatic phonographs which play the records—one at a time—into the telephone lines, through amplifiers (line, a.v.c., etc.). The program is sent to each location over the telephone lines, and at the location there is a suitable power amplifier and a speaker (or speakers) to reproduce the music.

Two large wired-music groups are spreading over the nation (and in foreign countries). These are World Music, Inc. and Muzak, Inc. In some towns and cities these two groups are quite active competitors. Each has a large library of suitable recorded music which is made available to their respective operator-licensees, according to the respective contract agreements.

Outside these two groups is found the independent operator who assembles his own library and program. Complete equipment is available for studio and remotes or "locations" from manufacturers specializing in this field.

A cross between the licensee and the independent is the sub-licensee, or sub-franchise holder who obtains his music by inter-city telephone lines from a licensee's studios in another city. In this case, the inter-city line is tied to the "locations" directly at the telephone company and the sub-licensee has only an amplifier and speaker in his office—like any other of his locations.

Actually, who uses wired music? What types of businesses are prospects for such a service? It is almost easier to list types of businesses who are *not* prospects! To name a few wired-music users, there are industrial plants, restaurants, department stores, hospitals, drug stores (and most every other type retail store), banks and similar institutions, professional men's offices (doctors, dentists, osteopaths, etc.).

To illustrate how diverse are the

businesses which use wired-music, let us give Telemusic of Stamford, Conn., as an illustration. Opened only a few months ago as a sub-franchise holder of World Music and getting their music by inter-city telephone line from Telemusic of Hartford, two enterprising local operators, Jerome Lambert and Herbert Chacon, have introduced their service to many varied Stamford locations. These locations include such diverse accounts as a department store, a just-opened Syl-May drug store (*see illustration*), Sorensen and Co. (manufacturers of electronic voltage regulators), a doctor, a men's furnishings store, a grocery and meat market, a bank, as well as several local restaurants and other businesses.

Something should be mentioned regarding the type of program offered such a diverse group of users. In the main it is a fairly quiet type of music. No vocals are used at any time. Both popular and classical music are included, but the popular music is never over-rhythmized and the classical music is never "heavy". A careful musical balance is maintained so that the program at each location creates a pleasant background against which business can be transacted.

**Equipment**

A typical amplifier for use with wired music is the Langevin Type 108-C. As may be noted from the schematic (*Fig. 1*), this unit employs three amplifying stages, contains a dual input, and has a power output of 20 watts. It employs six tubes, a 1612 (or selected 6L7), two 6SJ7s, two 6L6Gs and one 5U4G. It affords a gain of approximately 102 db from a low impedance 30- or 250-ohm source, a gain of approximately 61 db from a 600-ohm source and a gain of approximately 42 db when bridging a source with a nominal impedance of 600 ohms.

These inputs allow the amplifier to be operated from (1) a low-level, low-impedance source (such as a microphone), having an impedance of from 10 ohms to 250 ohms, (2) a medium-level medium-impedance source (such as a 600-ohm telephone line) having an impedance up to 1000 ohms, and (3) as a bridged device across a medium-level, medium- or high-impedance source (such as a 600-ohm telephone line) having a working range up to 25,000 ohms.

The output of the amplifier works into a nominal load impedance of 8 or 500 ohms. The output power of this unit is approximately 20 watts (plus 43 dbm) with less than 3% total harmonic distortion at 400-cycle single frequency. By reducing the plate voltage, the life of the tubes can be prolonged, and the unit can deliver 15 watts at less than 3% total harmonic distortion at 400-cycle single frequency. No intermodulation distortion figures are available at present because the industry has no standard method of making such measurements. The output noise is approximately 55 db below plus 43 dbm (12 db below .001 watt) when a 1612 tube or selected 6L7 tube is used.

The frequency response of the Langevin Type 108-C is plus or minus ¾ db to 1½ db from 30 to 15,000 cycles. The unit draws 150 VA maximum at 120 volts.

Negative feedback is introduced from the plate of one of the 6L6 tubes to suppressor grid and cathode of the previous tube (6SJ7), and from the plate of the other 6L6 tube to the screen grid of the same 6SJ7 tube.

Note also the cathode circuit of the two 6SJ7 tubes in which a portion of the circuit is not by-passed for degenerative purposes.

Paraphase phase-inversion is used wherein the grid of the second 6SJ7 tube is operated from the signal voltage across a resistor in the common 6L6 grids to ground circuit.

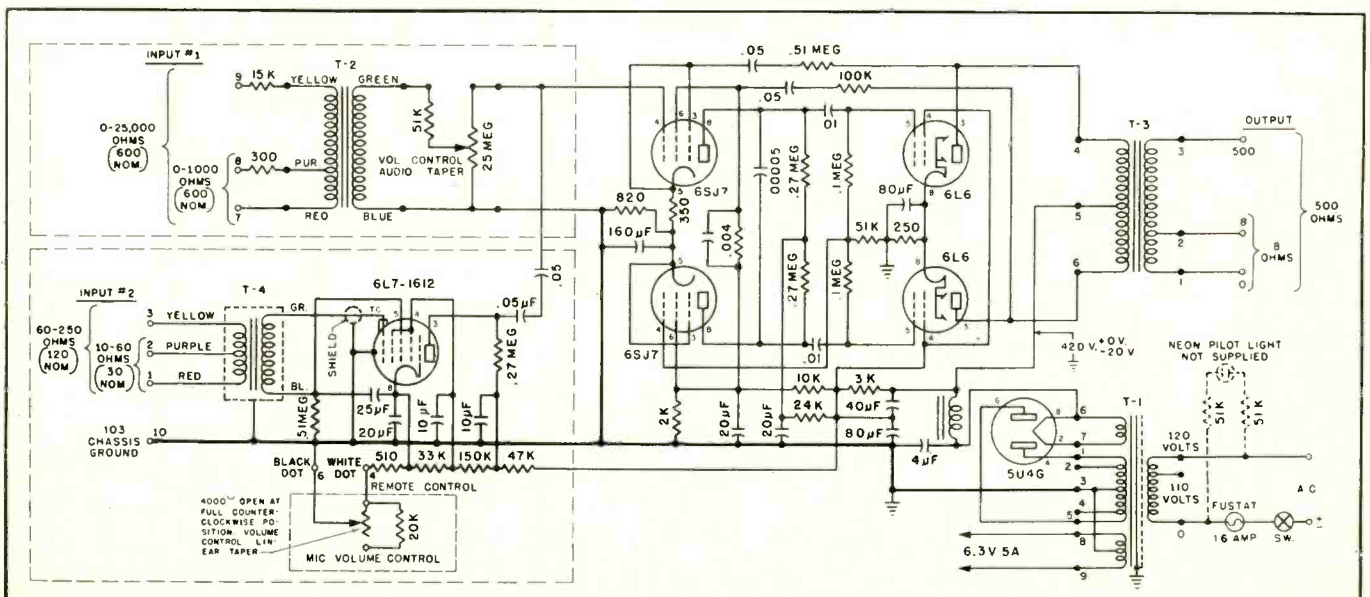
The gain of the medium-gain channel is adjusted conventionally by means of a 250,000-ohm potentiometer in the input transformer secondary circuit. The gain of the high-gain channel is adjusted in the cathode circuit of the 6L7 (or 1612) tube. An interesting feature is that this control may be moved to a remote location, or it may be paralleled by a 4,000-ohm linear taper potentiometer with an open on the full counter-clockwise position. In this case, the two potentiometers should be shunted by a 20,000-ohm fixed resistor and whichever volume control is not in use should be left in the "Off" position.

Due to the extended frequency response of the unit, all precautions should be taken to isolate input and output circuits. A ground is available on the unit for connection to earth. All output wiring should be run in twisted pair, or should be shielded.

**Other Markets**

The future of industrial music of the wired variety looks very bright. It fills a definite public service and creates for the businessman a more pleasant atmosphere in which to transact his business. Properly chosen music soothes the minds and nerves of patients in doctors' and dentists' waiting rooms, makes the repetitive work in an industrial plant less boresome. It provides a powerful new entertainment medium for all the people.

Fig. 1. Schematic diagram of the Langevin Type 108-C amplifier




**In this department the author, who is a very well-known record critic, will review monthly record releases of outstanding technical, as well as musical, quality.**

**EDWARD TATNALL CANBY**

In recording music, engineer and musician often work together closely. An interesting example of the composite kinds of knowledge they work out between them is the relation between musical dynamics and the dynamics of the VU meter.

The actual dynamic range of recorded music is greatly limited as compared with the natural range of the live music. In fact, so great is the degree of compression that the engineer may well pause to wonder why the recorded result is as musically satisfactory as it seems to be. There is at least one reason that he may not have thought of. A musician, with a little thinking of his own, will note that there are two factors at least in musical dynamics. One, of course, is the physical volume range, from loud to soft. The other is a pyschological factor that to my knowledge is not often taken into account, an effect of *apparent* dynamic range. The musician will admit that the greatest musical difference between "loud" and "soft" in most musical instruments is *not* volume—but tone color. "Loud" notes are far richer in overtones, "soft" tones lack them. In the case of some instruments the difference is so great that many listeners do not even realize the same instrument is playing them.

It follows then that since the musical *effect* of dynamic change depends on tone color as well as on actual volume, a recording which allows the listener to hear very clearly the changes in *tone color* that occur in a musical crescendo will very largely offset the cramping effects of dynamic compression. To put it another way, the greater the tonal range of your recording, the greater is the *apparent* dynamic range, given exactly the same compression. Here is an unlooked-for argument in favor of high fidelity recording!

There are interesting ramifications of this idea that lead off into music history (where this department is not prepared to go). Consider one more point, the gradual dynamic change versus the sudden. In music the most dramatic effect of all is the sudden "explosion"—where the extremes of the dynamic range are brought into closest contiguity. On records, however, this effect is perhaps the weakest of all dynamic changes. The electrical circuit is poor at sudden adjustments, and the human being, the engineer at the controls, is peculiarly at the mercy of these dreadful spots in music! Moreover, if the change is compressed successfully, the musical effect is still disappointing. Coming so rapidly, the compression is decidedly noticeable to the listener. And worse, the compensating effect of changing tone color mentioned above does not have time to operate, and thus that musical value is lost, too. The gradual crescendo or diminuendo, as we have seen, is another story. Here the compression, spread over a considerable distance, is much less noticeable, and the dynamic effect of tone color change is at its best.

All that remains for the curious engineer is to look into music history to find who wrote the violent dynamics and who preferred the long, gradual swell. Though it is beyond the scope of our discussion, there are in fact some simple and clearcut conclusions to be reached historically. Two examples will suffice here. The period of Beethoven, the early 19th century, is the worst for all-over dynamic violence! There has never been an adequate recording of Beethoven's 5th Symphony, no doubt for this very reason. Schubert's 9th Symphony (below) is similar. On the other hand, after 1850 or so the long swell-and-die-away type of music became overwhelmingly popular—and Wagner is its most extreme exponent—hence the highly satisfactory effect of Wagner's orchestral music on records. This type of music allows the maximum amount of tone color dynamics, to offset the compression of actual dynamics.

On almost any of the records below the effect of high fidelity tone color dynamics may be noticed by the curious listener.

Recent outstanding recordings for the audio engineer: (Number of records in parentheses)

**Virgil Thomson,** The Plow that Broke the Plains, Hollywood Bowl Orchestra, Stokowsky ........ **Victor M 1116** (2)

Apparently a new style recording from Victor—splendid high fidelity job. This is clever and good film music.

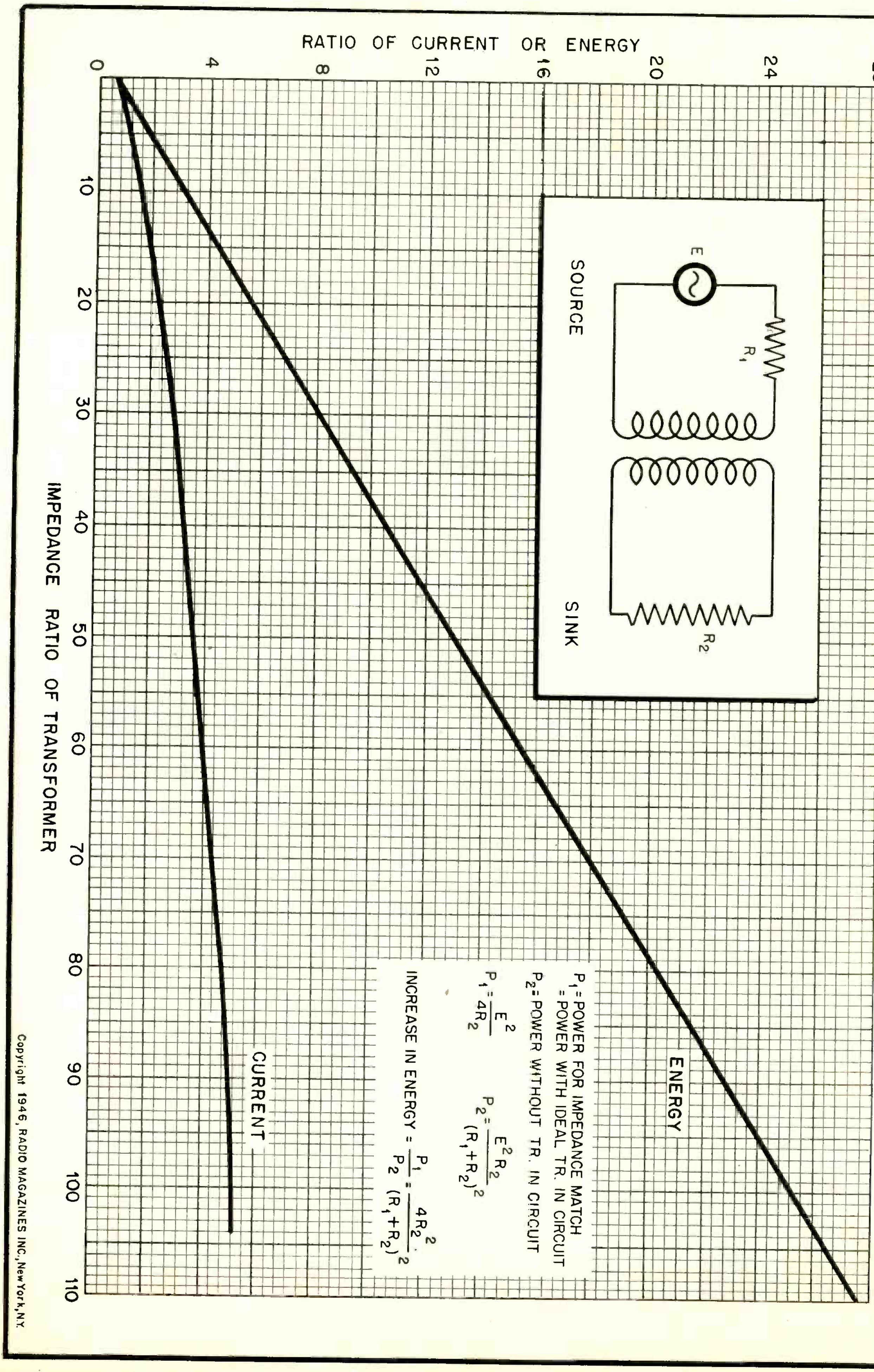
**Dumas,** The Three Musketeers (adapted by R. Rose). Errol Flynn, orchestra, etc. ............ **Columbia M 659** (4)

Done like a radio program, but the strikingly wider range recorded on this, as compared with usual AM broadcast, makes it technically interesting. Strictly radio-movie stuff.

**Schubert,** Symphony #9 in C Major. New York Philharmonic, Bruno Walter Cond. .......... **Columbia M 679** (6)

[*Continued on page* 40]

# AUDIO DESIGN NOTES




INCREASE IN CURRENT AND ENERGY DUE TO INSERTION OF IDEAL TRANSFORMER BETWEEN SOURCE AND SINK

# TECHNICANA

## BROADCAST STUDIO DESIGN

● Since the general evaluation of the suitability of a broadcast studio is based on the opinion of radio listeners, the performers in the studio, and others concerned with the production of programs within that studio, it is desirable to obtain a basis for measurements which may be thereafter related to the actual auditory acceptance. An analysis of the methods of securing this information and, in particular, the information necessary is the subject of an article entitled "Broadcast Studio Design" appearing in the May issue of *J. Acous. Soc. Am.*, under the authorship of H. M. Gurin and G. M. Nixon, both of NBC.

The problems of studio design are necessarily identical irrespective of the ultimate use of the studio for either AM or FM broadcasting, since the main object is to provide a suitable space for the actual microphone pickup. The best criterion by which a studio may be judged appears to be a study of the reverberation time with respect to frequency and the volume of the room. A flat over-all reverberation time vs. frequency curve appears to be the most suitable, with the major difference in opinion being concerned primarily with the region below 1000 cps. Above this frequency, the ideal curve should be flat, but a droop above five or six thousand cps is unavoidable due to air absorption, with humidity having considerable effect upon the upper registers.

As with most acoustic problems, the final consideration is how it sounds to the microphone, and common practice has indicated that room proportions of 2:3:5 are generally satisfactory, although in large studios it may often be difficult to obtain adequate ceiling height. The general-purpose studio, as contrasted to those used primarily for speakers or for audience shows, needs a higher reverberation time, and the space should be treated as a whole, rather than in separate areas. Microphone placement can then be relied upon to obtain the desired result.

In summation, it is shown that the reverberation time vs. frequency curve is important, but the acoustical engineer must use tolerance in its application. The frequency characteristic should be checked at various positions throughout the entire studio; means for controlling reverberation are considered necessary. Air absorption above 5000 cps limits the flatness of the reverberation time vs. frequency curve, and, in general, the all-purpose studio should have somewhat greater volume than indicated by normal volume vs. acoustical occupancy conditions.

## PRE-MODULATION CLIPPING

● Articulation tests conducted by the Psycho-Acoustic Laboratory, Harvard University, in 1943-1944 to assess the advantages of pre-modulation speech clipping in AM voice transmitters have been summarized in an article by K. D. Kryter, J. C. R. Lickliden and S. S. Stevens in the January 1947 issue of the Journal of the *Acoustical Society of America*. The results indicate that as much as 14 db in carrier power can be saved without loss of intelligibility if the speech is first subjected to 24 db of peak clipping and then reamplified by 24 db to obtain 100 per cent modulation.

The circuit of the speech clipping amplifier is shown in *Fig. 1*. Three advantages result from this procedure: (1) The effective range of the transmitter when using pre-modulation speech clipping is greatly increased, (2) over-modulation is permanently prevented, and (3) interchannel interference due to over-modulation is practically eliminated. The third factor may be still further improved by the introduction of a low-pass filter between the clipped and the speech amplifier.

The articulation tests indicate that to maximize intelligibility this method should not be used where the microphone picks up large amounts of ambient noise, although if transmission is impaired by atmospheric or electrical interference, speech clipping will result in a considerably easier to read transmission.

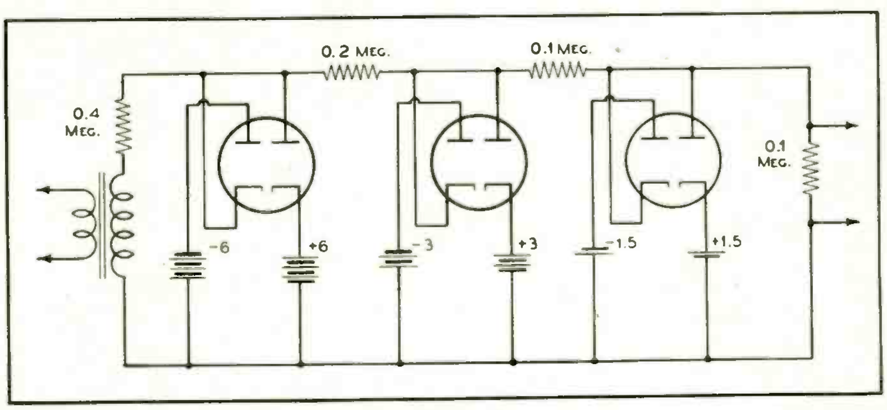


Figure 1

## ACOUSTIC CONSTANT and "LIVENESS"

● When an observer is listening to an orchestra or other sound source in an enclosed space he is aware of the direct source and the sound reflection from the walls. This reflection has a modifying effect which is commonly termed 'liveness.' A correlation between the 'liveness' and the acoustic constant has been developed by J. P. Maxfield and W. J. Albersheim and is analyzed in the January 1947 issue of the *Journal of the Acoustical Society of America*.

The authors find that the liveness constant *L* represents a statistical relationship of the physical factors which are fundamental to the process of hearing. The constant *L* is correlatable with the acoustic properties of the enclosed space and particularly with the distance between the source and the listener. When the relationship is reduced we find that the constant is also a ratio of the time integral of the energy density of the reflected or reverberant sound to the unintegrated density of the direct sound. Therefore, a time integral phenomenon in the human ear appears to be a fundamental factor in the interpretation of 'liveness' There are indications that several of the subjective effects which are functions of the 'liveness' are not fully accounted for by present theories based on steady-state conditions.

The validity of the use of the constant has been substantiated by empirical data. Although considerable work remains in this field, a method of calculating the constant based upon the subjective effects is presented, typified by the following:

(1) A change in the general tone

[*Continued on page* 47]

# NEW PRODUCTS

### ROBINSON PROFESSIONAL RECORDER

A new professional-type recorder has just been announced by the Robinson Recording Laboratories, 2033 Sansom St., Philadelphia, Pa. This instrument in-

corporates many features which the manufacturer has developed during their eighteen years' experience in professional transcription recording.

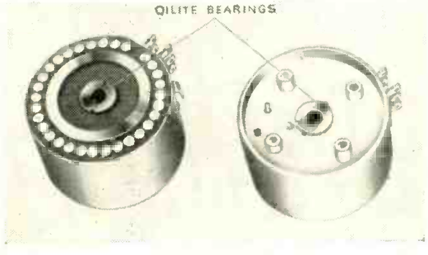
The equipment is designed for rapid operation together with absolute precision in results. A simple attachment provided with the recorder permits highly accurate eccentric and safety grooves when cutting master blanks.

For further information, please write the manufacturer.

### DAVEN ATTENUATORS

The Daven Company of 191 Central Avenue, Newark, N. J., announces another improved feature in its line of attenuators.

Oilite bearings are now being supplied on standard units. Two such bearings are provded on each unit . . . one at the switch end and the other at the shaft or detent end.

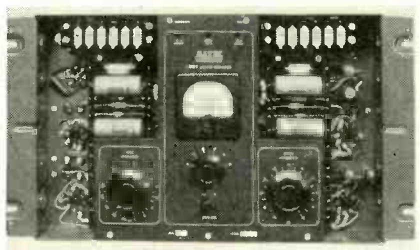



The superiority of this type of bearing is due to the fact that it is made of an oil impregnated metal. Because of the inherent characteristics of oilite, the bearings are permanently lubricated, and during the life of an attenuator in normal service, no oiling or greasing will be required. To the operator of an attenuator with oilite bearings this means a free-turning, nonbinding unit.

### LIMITER AMPLIFIER

The Altec Lansing A-322C Limiter Amplifier which has been recently announced has several outstanding features which make it valuable for use in film and disc recordings, broadcasting, and public address applications. It has rapid attack time, automatic volume control, high power output and limits with less thump. Its use in film and disc recording prevents overmodulation percentages without the danger of exceeding modulation limits. In public address systems it will adequately compensate for variations in volume resulting from changes in the speaker's position with respect to the microphone. Its use will prevent blasting and will allow the use of a higher sensitivity in the p-a system without feedback. In high-powered public-address systems, the use of the limiter will prevent sudden or accidental peak volume from causing loudspeaker failure.

This limiter amplifier operates as a true linear amplifier up to a specified level and then limits the volume output beyond



this point. The signal circuit consists of three balanced push-pull stages. The input stages use type 1612 variable mu tubes; the second stage is resistance coupled, using type 6J7 pentode tubes, and the output stage uses type 6V6GT beam power tubes. A pair of cathode followers (6SN7GT) bridge the plates of the output tubes and feed a dual diode (6H6-GT) which rectifies the control voltage that is applied to the type 1612 tubes. The output of the cathode followers appear as a low impedance charging source resulting in a very rapid "attack" time.

A meter having a "100%" point and calibrated in DB Limiting permits the operator at all times to know the amount of limiting taking place. When desired a remote DB Limiting meter can be connected. Provisions are also made so that the N.A.B.-orthacoustic pre-emphasis equalization which is a built in feature can be utilized to good advantage, since the limiting is controlled from the pre-equalized voltage.

With proper balancing, the "thump", when limiting, will be 50 db down from the signal level. This is accomplished through the use of well-balanced transformers and other components.

#### Specifications

Gain:
1000 cycles; Equalizer out: 68 db. 1000 cycles; Equalizer in: 60 db.

Hum and Noise:
Equalizer out: -37 dbm -45 db. Equalizer in: -44 dbm -52 db.

Frequency Response:
Equalizer out: ±1 db 20 to 20,000 cycles.
Equalizer in: (N.A.B.-Orthacoustic Recording Equalizer).

Limiting Threshold:
Normal: +25.2 dbm; +17.2 db. High: +33.2 dbm; +25.2 db.

Power Output: 5 watts.

Limiting Ratio: 10:1.

Attack Time: 0-.0002 sec. (0.2 milli-second)

Release Time:
Normal 0.5 second (can be varied by changing value of one resistor).

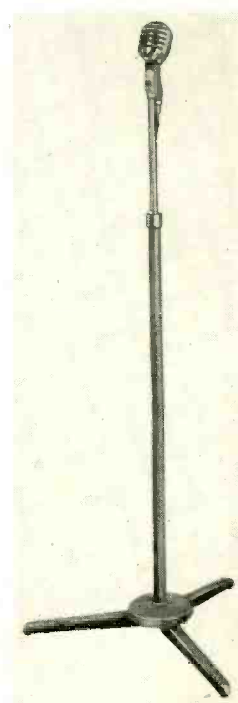
Input Impedance:
600 ohms; 30 db; 30 step attenuator.

Output Load Impedance:
600 ohms; 20 db; 20 step attenuator.

Power Supply Required:
275 V.D.C. Regulated @ 100 ma. 6.3V. 2 amps. (Use Altec Lansing Regulated Power Supply.)

### MIKE STAND

A new utility model 430 button-control floor stand is announced by Electro-Voice, Inc., Buchanan, Michigan, as a companion to the E-V deluxe model 425 floor stand. This new stand offers many unique features not available before at such moderate cost. A single red button gives instant finger-tip control of shaft height. You simply press the button with the finger, and easily raise or lower the extension shaft with the other hand. Release of red button automatically locks shaft firmly in any position.



For further information, write Electro-Voice, Inc., Buchanan, Michigan. Ask for Bulletin No. 134.

## NEW V-T VOLT-OHMMETER

Television type v-t volt-ohmmeter. Reading to 15,000 volts d c., with d. c. ranges 3 - 30 - 150 - 300 - 600 - 3000 - 15,000

volts. A. c. ranges, 3 - 30 - 150 - 300 - volts, good to 300 megacycles. Ohms: 1000 - 10M - 100M - 1 Meg - 100 Meg.

Careful construction includes heat loop on all critical resistors. Fungus proofing to maintain stability and accuracy over a long period of time. High d. c. voltage reading and high frequency a. c. readings that are necessary in servicing television receivers are available in this instrument. Also suitable for servicing regular receivers, F.M., transmitters, and other equipment.

For further data write Electronic Manufacturing Co., 140 So. 2nd St., Harrisburg, Pa.

## KNIGHT AMPLIFIER

Allied Radio Corporation announces a new 4-watt phono-amplifier, specifically designed for record-playing purposes. The unit has an inverse feed-back circuit and 4 watts of output which easily provide the drive necessary for an 8" or 10" speaker. It operates with any high-impedance crystal pickup and 110-volt phono motor.

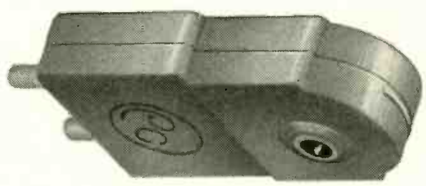
The amplifier is small, measuring only 4" wide, 5¼" long, and 4¾" high with tubes inserted and so can be combined with a speaker and turntable in a small cabinet as a complete record player.

For complete information write to Allied Radio Corporation, 833 West Jackson Boulevard, Chicago 7, Ill.

## PICKERING CARTRIDGE REPRODUCER

Bringing the advantages of the Pickering Pickup to the user of conventional record players and changers, the Pickering Cartridge is now available at most distributors. The moving system which generates the electrical signal is identical in construction to the Pickering Pickup which has been so successful in the highest quality reproduction of phonograph records and lateral transcriptions.

A unique mounting, the keystone clip, attaches to any standard pickup arm, and the cartridge slides firmly onto the clip, permitting longitudinal adjustment for minimum tracking error. The cartridge has a sapphire stylus with a tip radius of 0.003 in., considered ideal for reproduction of shellac records. The output, at approximately 50 mv, is designed to be fed to the grid of a tube.

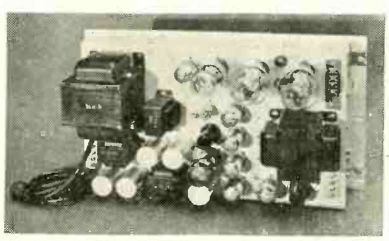


A simple circuit provides a 12-db/octave drop-off above 4,000 cps, when desired, for reduction of surface noise.

Further information can be obtained from Pickering & Co. Inc., 29 W. 57th St., New York 19, N. Y.

## UNITIZED AMPLIFIER SYSTEM

Fairchild Camera & Instrument Corp., has announced the Unitized Amplifier System for the varied requirements of the sound recording industry, ranging from the simplest portable sets to the extensive multiple channel installations in professional recording studios.



The dozen or more units comprising the system are matched for easy mounting on the customer's rack. The group can be expanded at will to provide 30 or more different combinations without obsolescence of any basic component. If a portable amplifier is desired, it is furnished by assembling the needed components into a special trunk. These arrangements provide the utmost flexibility in multiple channel amplifier systems, and for many other applications in recording.

The unitized amplifier system consists of a total of 12 units. Of these, the No. 620 power amplifier and No. 621 microphone pre-amplifier are now ready for delivery. Additional units to come out shortly are pre-amplifiers and boosters for pickup and line; output switch panel with volume indicator and monitor take-off; input switch panel; N.A.B. and variable equalizers; mixer; volume indicator panel; bridging device; and, auxiliary power supply.

Inquiries should be addressed to C. V. Kettering, Fairchild Camera & Instrument Corp., 88-06 Van Wyck Blvd., Jamaica 1, N. Y.

## REK-O-KUT CONSOLE

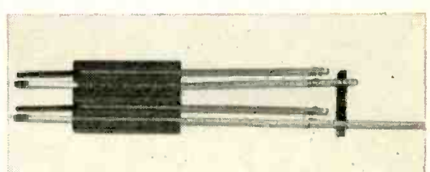
A beautiful new console (transcription or recording cabinet) is being shown by The Rek-O-Kut Company, 146 Grand Street, New York City. This cabinet is

sturdily built, being made specifically to mount the various Rek-O-Kut recording and transcription turntables. It has a drop front door and self-contained pockets for holding approximately 100 sixteen-inch records. The console is finished in a metallic two-tone grey—dimensions 32" high by 24" wide and 26" deep. Outlets and terminal blocks for motor line and pickup are mounted on the motor board. Four screw jacks are provided for leveling the console. The entire unit can be sold separately or in conjunction with any of the regular Rek-O-Kut transcription or recording turntables.

## CONTACT SPRING ASSEMBLIES

P. R. Mallory & Co., Inc., has just released a new and unique specification sheet on contact spring assemblies. It contains detailed dimensional drawings of three groups of standard contact springs, permitting a selection for conventional spring stack-up in a wide range of circuit possibilities.

In addition to solving standard contact asembly problems, use of this specification sheet permits selection of many proprietary alloy and contact materials to meet special electrical contact problems.



This specification sheet is printed on tracing paper suitable for blue printing. Copies of the specification sheet may be secured on request from P. R. Mallory & Co., Inc., Indianapolis 6, Ind.

## Record Revue

[*from page* 34]

One of most difficult symphonies to record, thanks to violent dynamics (see above)—but this is a fine job, and musically a superb reading. Wide range recording.

**Blitzstein,** Symphony, "The Airborne". New York City Symphony, RCA Victor Chorale. Leonard Bernstein; Charles Holland, Walter Scheff, Robert Shaw, Narrator ............ **Victor M 1117** (6)

This marks the coming-of-age of the radio technique of orchestra, narrator, chorus, as applied to a symphonic work. Not great music, but one of the most exciting dramatic pieces ever put on records. A marvel of technical coordination; interesting effects of varying liveness.

## Limiting Amplifiers

[*from page* 19]

amplitude by the limiting amplifier. Thus overswing will be prevented.

However, if a large amount of limiting is employed in a pre-emphasized system, it will be observed that the program frequently would suffer changes in loudness without apparent good reason. This is the result of the pre-emphasis increasing the intensity of high-frequency components of the program material to such a level as to necessitate gain reduction. Thus level control becomes tied up with the frequency as well as with the amplitude of the program material, whereas, in a flat system, such as conventional a.m., level control depends upon amplitude alone. Thus, in a pre-emphasized system, limiting must be held to low values, such as 2 or 3 db; otherwise it will become objectionably noticeable to the listener. Since the limiter is to be located after the pre-emphasis, it serves primarily as an overmodulation guard and allows only a nominal amount of signal strength increase.

Good practice in f.m. would therefore be to adjust the amplitude of the flat portion of the compression characteristic so that ±75 kc swing is obtained at some high audio frequency. The program would then be applied to the limiting amplifier and the input level set so that only occasional 2 to 3 db gain-reduction kicks would be obtained with normal program levels as observed on the station VU meters. The slight amount of gain-reduction observed should occur on program material rich in high frequencies.

## Planning a Studio

[*from page* 9]

in increments of 1¾ inches, the same increment used for standard panel sizes. Trial rack layouts can be made rapidly to scale by use of this device without the necessity of adding the fractional dimensions of panel heights to determine if everything will fit in the rack. Another very useful aid to rack layout is a piece of one-inch wide masking tape 77 inches long, marked off in inches and put up on some convenient wall. This tape so marked off provides a dummy rack which will give the engineer standing in front of it an idea of the proper height for various components such as jack fields, meters, etc. (The tape should start 4 inches from the floor since most standard racks have a four-inch kick plate at the bottom.)

It should be noted that jack fields never come out to standard panel dimensions due to the fact that jack strips are not an increment of 1¾ inches. In the case under discussion where the field is made up of four double and one single strip, the over-all height is 9¾ inches. The nearest standard panel size of rack space necessary to accommodate the field is 10½ inches. A ¾-inch-panel must be used to make up the difference. The location of mounting holes on the rack prevents placing of standard panels immediately adjacent to the odd size of 9¾ inches.

The following simple rules will aid in the actual rack layout.

1. The jack field should lie between 40 and 55 inches of rack height. If lower, it is difficult to read jack designations. If higher, it is hard to insert a plug.
2. All meters should be at eye level.
3. In general, low-level equipment such as preamplifiers should be mounted near the top of the rack. Monitor amplifiers should be in the lower portion of the rack.
4. Amplifiers having gain controls should be located so the controls can be reached without stretching or stooping.
5. When possible, avoid placing any equipment having controls immediately below a jack field. Dangling patch cords can get in the way.
6. Allow at least 14 inches of blank space at the very bottom of the rack to accommodate terminal blocks.

*Fig. 3* shows the rack layout arrived at for the studio under discussion. The line coil, the two auxiliary transformers, and the two divide pads were mounted on left-over space on the shelves mounting the monitor and cue amplifiers. As racks go, this one ended up rather crowded, with the jacks being slightly on the low side. However, it is sometimes impossible to do better without causing a greater inconvenience elsewhere.

[*To be continued*]

## Musical Acoustics

[*from page 27*]

circles represent the troughs. The amplitude of a wave is lessened as it travels radially from the source. Only on the perpendicular bisector of the line joining two sound sources of equal frequency and intensity are the amplitudes of the waves from both sources equal and of the same phase, thereby producing a combined wave of maximum amplitude, double that of either source wave. To either side are paths of waves of minimum amplitudes where crests meet troughs. Other nodal zones of superimposed crests on crests, and crests on troughs, of other resultant amplitudes are shown by solid and dashed divergent lines of a hyperbolic nature. On them the wavelength measures longer, so the frequency appears less for the interference waves which travel outward on such lines without destruction of the entity of their component waves. Actually, two sine waves of this same frequency combine at any point in a medium of linear transmitting properties to give a wave frequency identical to the component waves, so graphical representation promotes a fallacy. See Part III.

Similar interference of sound waves may occur with reflected waves from a single sound source. *Fig. 4* represents the plan of an auditorium 60 feet long by 40 feet wide, having the sound travel at a velocity of 1120 feet per second and originate centrally 12.5 feet from an end wall. The arcs represent the location of wave fronts one-thirtieth of a second after the origin of the first sound, with reflections from the rear and the sides before the direct wave has reached the front end wall. One can imagine how confused the picture of wave interference would have been only one-tenth of a second after its origin. Reflections assure about the same average of loudness in all parts of the room. But if the length of travel of a reflected sound, because of the size of the room and good reflection from its walls, cause it to reach the ears of an auditor more than one-fifteenth or one-twentieth of a second after the direct sound reception then disagreeable, confused reverberation (echo) occurs.

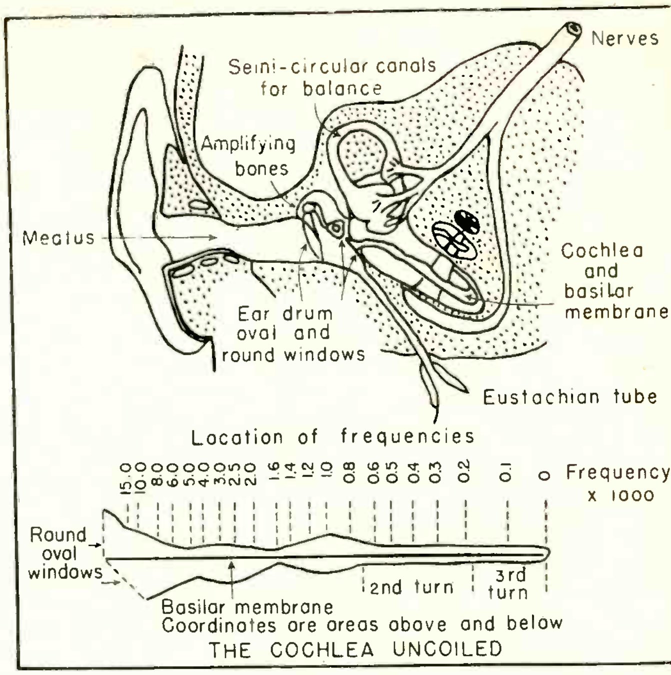



Fig. 5. Diagram of human ear.

The change in frequency as well as amplitude from interference of reflected waves, as shown in *Fig. 3*, may cause us to wonder why anything but dissonance is heard from music in a chamber, unless the ear and brain first analyze and then synthesize sound stimuli into tonal patterns. Therefore, we must consider how the mechanism of the ear affects auditory perception.

The ear has three chambers:—the outer ear canal (meatus) which terminates at the ear-drum (tympanum); the middle ear inside the ear-drum with a lever system of three tine bones (hammer, anvil, and stirrup or stapes) which multiply the pressures 30 to 60 times and transmit them through the oval window to the fluid in the inner-ear (the cochlea), a bony cavity with two and three-quarter spiral turns like a snail shell, which is divided by partitions into three long narrow chambers throughout its length and carries the basilar membrane with its 20,000 or more "rods of Corti", each with its twelve or more cilia. The basilar membrane with its rods respond to various frequencies, with the higher frequencies perceived nearer the oval window, all as depicted in *Fig. 5*. From the bases of the "rods of Corti" auditory nerves

[*To be continued*]

# RECENT AUDIO PATENTS

### Enhancing Sound Reproduction

● It is well known that if a listener hears a group of musical instruments playing in an enclosed room he will hear only a very small portion of the total sound as direct radiation. Probably 90 per cent of the sound energy reaches the ear through reflections from the walls and other indirect paths. A method of artificially enhancing sound reproduction was patented May 13, 1947, by C. M. Sinnett.

In the system proposed by the inventor, reproduction is based upon the *binaural* characteristics of human ears. This is accomplished by providing synthetic reverberations through the use of a plurality of sound reproducers which are spaced in a predetermined relation with respect to the listener. The reproducers are supplied with audio-frequency waves through separate channels, and at least one of the channels is provided with dynamic range expansion.

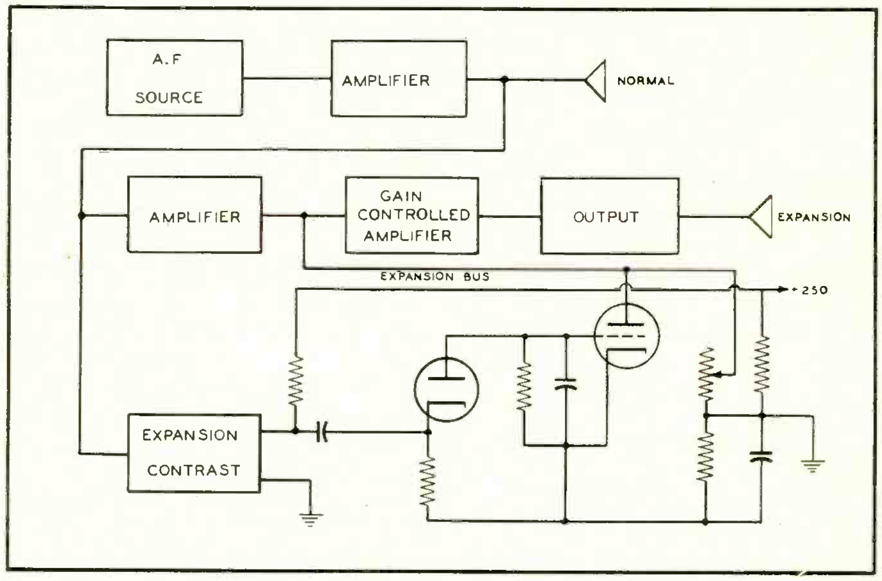



Figure 2

*Fig. 1* illustrates the basic idea of this invention. The output of the turntable is fed into the normal amplifier and the expander amplifier. The purpose

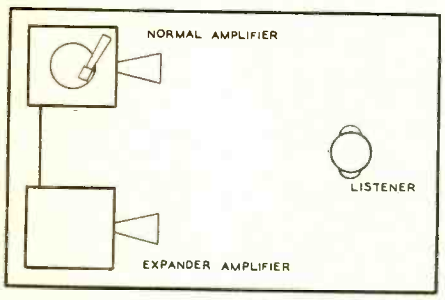



Figure 1

of the expander is to heighten sound perspective for the listener. This is accomplished by arranging the output of the expander amplifier to provide a reverberant "build-up" time. *Fig. 2* shows the arrangement of the individual components of this system. The expander operates in the usual fashion by applying the rectified output voltage of the diode to a direct current amplifier triode. Since this triode is effectively operating with *B plus* to ground, the voltage drop in the plate resistor results in a negative signal output with zero audio signal input to the diode. An expansion of about 14 db with an input of 0.5 volt can be obtained in this fashion.

The preferred method of utilizing this system would be to enhance sound reproduction by dividing the normal amplifier output through a system of filters. The output of each filter is then fed into a separate expanding amplifier and separate reproducer. The division of the audio frequency spectrum might be on the basis of 30 to 120 cycles; 120 to 500 cycles; 500 to 2000 cycles; 2000 to 5000 cycles and 5000 to 10000 cycles. This plurality system would reproduce a complete musical selection according to the audio-frequency energy content of each individual channel.

The patent, No. 2,420,204, is assigned to the Radio Corp. of America.

### Sub-Audible Indicator Circuit

● An indicator circuit for providing an audible indication of a signal which has a frequency outside the normal audio range has been patented by H. J. Woll. Since it is frequently desired to detect audio notes which cannot be appreciated by the senses (viz., one to five cycles per second) this invention should prove to be very useful in certain applications.

Design of the circuit is illustrated in *Fig. 3*. The sub-audible input signal is fed into a 1L4 type tube which serves as an amplifier of the low frequency note. The output is coupled to the control grid of another 1L4 which serves as a balanced type detector. The third 1L4 is used as an audio oscillator with an output of about 1000 cycles. The audio oscillator output is coupled to the

[*Continued on page* 44]

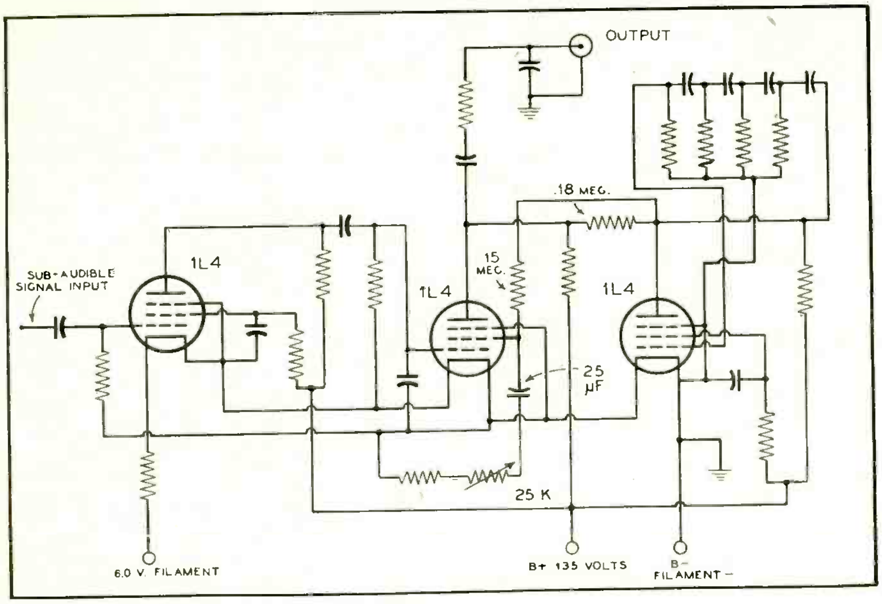



Figure 3

screen grid of the 1L4 detector through a 150,000-ohm resistor, which is also connected to the filamentary cathode circuit through a .25 µf condenser and a variable resistance network. This network prevents any phase shift in the audio signal. The audio oscillator is also coupled to the plate circuit of the detector tube through a 180,000-ohm resistor.

The operation of the detector tube may be explained by the nullifying effect of the two audio voltages applied to the screen and the plate. The 25,000-ohm variable resistor in the screen circuit is used to produce a complete null of the 1000-cycle note in the output circuit when no sub-audible input signal is applied. This tube is therefore operating as a normally closed electronic valve, preventing any of the 1000-cycle tone from reaching the audio output connection. If a very low frequency audio note is now applied to the amplifier tube, the amplified signal which now appears on the control grid will unbalance this stable condition in the detector tube and the attenuation in space current flow will allow a 1000-cycle note to be heard in the output connection. This circuit, although providing an audible indication of a sub-audible note, does not identify the frequency of the sub-audible note.

The patent, No. 2,420,404, is assigned to Radio Corp. of America.

### High Potential Voltage Regulator

● A new electronic type voltage regulator which enables the designer to use low voltage receiving tubes to control high voltages has been patented by the Australian inventors, Alfred J. Irish and Donald G. Lindsay.

The inventors claim that the principal reason why it has not been possible in the past to use receiving type tubes in the control channel is because it was

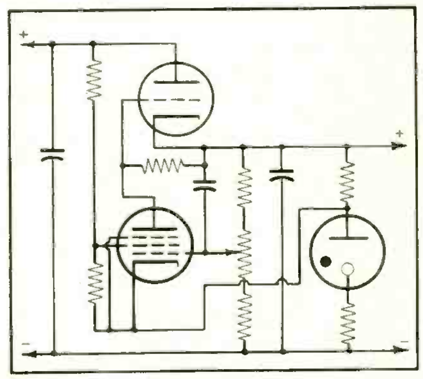


Figure 4

previously thought necessary to maintain the cathode of the amplifier at a fixed potential with respect to the negative conductor of the system in order to obtain good regulation. The system proposed by the inventors has been found to operate on any value of cathode potential if sufficient amplification is available in the system to compensate for any variation in the cathode potential.

The voltage regulator circuit is shown in *Fig. 4*. The pentode amplifier is operated with its cathode at a high potential, although the potential difference between the cathode and the plate permits the use of receiving type or low voltage tubes. Generally, this tube would be a high mu pentode with the screen and control grid voltages obtained from voltage dividers.

In operation the potential difference between the cathode of the amplifier tube and the high voltage lead is between 200 and 400 volts. A three-resistor series combination is employed as a means of obtaining the reflected load variations which are to be utilized by the amplifier tube. A resistor is connected between the plate of the amplifier tube and the cathode of the regulator tube. The variable impedance of the regulator tube operates in the usual fashion associated with this type of voltage regulator, i.e., a change in load to decrease the output voltage (greater current drain) also changes the grid potential of the amplifier tube (less positive), reducing the current flow through the regulating resistor coupled to the grid of the voltage regulator tube which in turn reduces the impedance of this tube and consequently raises the impedance of this tube and consequently raises the impressed voltage to the load.

The patent, No. 2,416,922, is assigned to Amalgamated Wireless (Australasia) Limited.

## Preamplifier Noise

[*from page* 16]

gram and noise. The reading is determined by the greatest deflections occurring in a period of about a minute, excluding not more than one or two occasional deflections of unusual amplitude. The flicker effect may appear rather noticeably at the lower noise levels. If both noise and program are read in vu as described, it would seem that at least a certain amount of allowance were being thus automatically provided for "peak factor"[10] (the assumed average of 10 db by which the peak value of program waves exceeds the indicated value). Therefore, measurements made as herein described might be expected to yield a reasonably just figure for operating noise.

For repeated tests, the oscillator setup shown in *Fig. 1* may be convenient in duplicating in a steady-state manner the volume reading obtained from the microphone. The oscillator output level is read on the indicator $VU_2$; pad 1 is a calibrated attenuator (such as a decade); pad 2 has an output impedance equal to that of the microphone internal impedance. The input section of a transmission measuring set will serve nicely for the attenuator and meter setup shown.

It will be noted that the internal noise level of the noise meter does not have to be as low as the first-stage noise of the amplifier being tested. The amplification contributed by the tested amplifier provides a margin for over-riding the noise in the noise meter.

Since published noise level specifications on FM transmitters for broadcasting are in the neighborhood of −65 to −72 db as compared with 100% (plus or minus 75 kc) modulation, it is seen that the preamplifier may well be the limiting factor in obtaining a system noise level within FCC tolerances. Preamplifier noise in the FM station will, of course, be evidenced as FM noise on the carrier. The preceding discussion does not relate to the limitation placed by the FCC on AM noise of the FM station; such AM noise is likely to have its source in the transmitter.

[10] H. A. Chinn, "CBS Control-Console and Control-Room Design," *Proc. I.R.E.*, Vol. 34, page 295; May, 1946.

## Ultrasonics

[*from page* 24]

run a series of curves-constant/speed variable pressure, also constant pressure/variable speed. For a given tuned resonant chamber, we find one or more resonant spots where the power handled is up to 6 kw per $cm^2$ at only 40 lbs air pressure, and 10 kw at 61 lbs. (The area is referred to the cross section of the orifice.) See dotted curve, *Fig. 4.*

The mass of air in the tuned cavity is easy to calculate. We accelerate to sonic velocity in less than five microseconds, which requires a pressure of nearly 200 $lb/in^2$ when we have 40 lbs static pressure, so we are evidently cavitating the gas in the cavity.

Calculation is most difficult for a case of cavitation. Also we all remember our physics course where we were told that a perfect gas has a fixed relationship between pressure, volume and temperature ($PVT=K$.) When working in microseconds, however, another important term enters, namely, time. If we instantly compress a gas, kinetic theory says there must be so many collisions between molecules to increase the temperature. The necessary number may take more than 5 microseconds to occur, and be still higher in some gases. So we need some new theory to cover this case, at least in combination with the other factors at such high intensity.

Observation shows that we discharge about 40% of the gas in a cavity per cycle. Integrating the velocity/mass, it works out that about one-third of the internal energy in the gas is in the form of kinetic energy. To recover the remainder we do as the turbine boys do — use a conical nozzle to increase the velocity by the expansion of the gas. We use no exponential horn, as this energy is not audio—that is, it is not a.c. yet. The nozzle is designed to increase the velocity to just under 1500 ft/sec, where theoretically all energy appears as kinetic.

This again interposes a difficulty. In radio, nothing can travel faster than the speed of light. Now, a wave traveling in an elastic medium at the speed of sound is a sonic wave in that medium. But a wave can be produced which travels faster than that, and when it exceeds sonic velocity it is a "shock wave". We are all familiar with the difficulties the aircraft boys are having in trans- and supersonic flight. They are working in the shock-wave region where phenomena occur that are poorly understood.

When we work this shock wave into another tuned pipe, as shown in *Fig. 5*, we have reached the end of current theory and must design by the Edison method. However, in many cases we shall pass the material we wish to treat through the machine itself, and the more work we do on it internally the better. Such a load would be breaking up gases, for instance, or subjecting particles borne by the gas to tremendous accelerations. That would be set up as a closed system, of course.

### Construction

Some hints on the construction. A nice rotor size is 3½ inches. Since we may want to rev up in excess of, say, 40,000 r.p.m., this should be an X-rayed cheese forging, upset from 2-inch stock. Nitralloy "G" is a suitable material. Eighty square $.062 \times .062$ cuts should be made around the periphery. These should be accurate to about a quarter-thousandth inch all around, dead true and concentric. It should clear the housing by no more than 1½ thousandths, and clear the stator by the same amount. If it passes through resonance on the way up and hits the stator, you have lost a rotor.

Rim thickness can be from 6 to 30 thousandths. Remember we must chop several million slugs per second if we use 80 jets on the stator, and a rotor requires power for driving. The thicker rotors may be given some angle in cutting the vanes so as to help carry the load — about twenty degrees will carry it at 15 h.p. of air, for instance.

The stator can have from one to 80 jets. For most work, try to get along with one jet as a fifty h.p. compressor is needed to run this little turbine to full capacity, plus some to spare. One jet can give useful effects with a half-horsepower compressor.

For a motor and spindle, we can use a good air grinder, or a compact type of tool post grinder, force the head on the wheel side, and grind everything dead true in a good mandrel in place of the spindle.

The stators are actually square gears with the teeth 250, 350 and 500-thousandths thick to resonate at odd spots from 18 to nearly 40 kc. Make the body some common dimension (½") and grind off the unused sections of teeth. It is worth while to lap the inside of the teeth. This is not a job for an ordinary machine shop — also remember it is one thing to grind a cup round to a quarter-thousandth and another to have it stay that way.

The stresses on the rotor are not too high, but there is a definite advantage in not using round holes to pass the gas, as they have large stress concentrations and tend to break loose just above the hole. By using the slot construction, this effect is eliminated.

### Using Steam

When using steam as a power source, it is not practical to work with such a small rotor diameter as three and one-half inches. It is necessary to use ball bearings to secure the close clearances desired both axially and radially. Since we can only hope to use steam with some superheat to prevent the erosive effects of wet steam, it is very difficult to cool the ball bearings with such a small rotor. It is suggested that a rotor diameter of at least 7 inches be used for steam. The nitralloy will withstand seven or eight hundred degrees F.

Since this is a gas-type generator, it is naturally adapted to gas loads. Is there any way of impedance matching to liquids or solids?

A good analogy of a sonic impedance match from gas to solids is to visualize a long metal bar, cold at one end and gradually becoming hotter until the far end is gas. This would be an impedance-matching transformer — we could put in our energy at any impedance and take it off at quite a different impedance, by tapping the bar at proper points. Another attack is to match two impedances by inserting an intermediate impedance which is a geometric mean between the two.

Attempts to bridge the enormous gap between the impedance of a gas and that of a liquid has been investigated theoretically. By choosing the heaviest gas and the lightest liquid, it was found that at 400 pounds pressure, the gas would have a density equal to the liquid. It is easy enough to seal the turbine to withstand 400 lbs. Very heavy gases would probably be broken up, though, if they were circulated in a closed system at high power.

The problem of liquid and solid loads can probably best be handled by going to some form of liquid generator instead of attempting to bridge the great gap between gas and liquids. We shall discuss some possibilities in this direction in the next article.

Some commercial uses of this device

might be mentioned. The Germans say dust can be precipitated by agglomeration at the rate of 1½ watts per cubic foot per minute. To compete with the Cottrell precipitator, super-heated steam power should be used. There is no fire hazard in coagulating explosive dusts. Various experimenters report agglomerates formed this way have very high electric charges and velocities when they hit a boundary layer, so their adhesive powers are extraordinary with some combinations.

Hydrocarbon chains can be broken up, and some products formed that cannot be produced any other way. The French found that a strong ultrasonic wave in mixed gases tended to separate them. These short waves inside an engine cylinder might produce sufficient extra pressure to produce Diesel operation without Diesel over-all pressure, and they travel 2,000 ft/sec in hot gas.

Some interesting effects have been noted in increasing the efficiency of combustion in a liquid fuel flame. We may be able to sterilize the air in any industrial place, in addition to precipitating the dust. If harmful bacteria in a baby ward can be made harmless by a million "G" acceleration, we can turn out a useful unit.

## Cathode Phase Inverter

[*from page 22*]

stage can be designed to give results comparable to the conventional class A push-pull type. In addition, it offers a reduction in tubes and components for the over-all amplifier, although the output transformer is of special design. The number of tube types usable is limited, however, because of the practical limitations on the value of $R_K$. Low plate resistance triodes, such as the 2A3, for example, yield a rather inefficient design for reasonable values of $R_K$.

### Some Conclusions and Precautions

The foregoing examples were chosen to illustrate the design of cathode phase inverters and should not be taken as representing optimum designs. Among the many applications[2] of this circuit is the d-c amplifier. This form has not been treated here, because the same general theory applies and the individual designer can best use it to suit his specific purpose.

In the design and use of this type of phase inverter several things should be kept in mind, if best results are to be obtained. Among these is the requirement that the tube parameters $\mu$, $R_p$ and $g_m$ must be known with reasonable accuracy. In the transconductance form of the design equations, $g_m$ is always the effective value as determined from equation 8, when the $g_{mo}$ of $V_2$ is known at the chosen operating point, or as determined from the load line when the operating point is considerably removed from the recommended. This is particularly true when working with triodes, but not so important with pentodes when the effective and rated $g_m$ can be almost the same.

The inverter stage can be designed to have equal plate loads, if $\beta$ is allowed to have some reasonable value other than unity. In this case equation 9 or 10 is solved for $R_K$, *with* $R_{L1}/R_{L2} = 1$ and $\beta$ equal to the assigned value. An alternative is to use curve *A* on *Fig. 1.* The allowable unbalance in per cent is decided upon and the corresponding value of $R_K$ read off the curve. This value of $R_K$ is then corrected if $g_{m2}$ is other than 1000 µmhos. For reasonable unbalance the value of $R_K$ will usually be rather high and the method of determining the operating point can be similar to that described for example number three.

Since the amount of degeneration is small it cannot be expected to contribute much to the reduction of distortion. In the designs that have unbalanced plate loads, the tube $V_1$ will be the limiting factor on output, because its grid swing will be larger than that of $V_2$. In this connection the designer will find equations 15, 16, 17 and 41 useful in determining the maximum output in terms of the allowable grid swing on $V_1$.

The equations and design procedures outlined herein have been closely checked by experimental procedure and use. Hence it is felt that, intelligently used, they will yield gratifying results. It has been shown how the cathode phase inverter can be used in a variety of ways. In certain forms it has considerable stability, yields adequate gain and offers economy of components. The latter is of considerable importance where cost is a factor.

[2] Cathode Phase Inversion, by O. H. Schmitt. *Review of Sci. Insts.* Vol. 12, Pg. 548, 1941.

## Technicana

[*from page 36*]

quality to improve the musicians 'resonance' or 'roundness'.

(2) Blending of the various instruments of the orchestra into a single coordinated sound.

(3) sense of acoustic perspective.

It is evident, that the constant must however, be interpreted as a statement of the average throughout the space and the frequency range.

# NEW CATALOGS

### AMPLIFICATION BOOKLET

This 24-page booklet interestingly describes 20 essentials for perfect amplification. Some of the unusual features covered include direct-coupling, scratch-suppression, increased musical range, signal-expansion, power requirements, noise, higher fidelity, extended dynamic range, "presence", hum elimination, distortion reduction, microphonics, response control, grid-current, delayed plate-voltage, fixed-bias, balanced audio signals, voice accentuation, reduction of thermal agitation, and cross modulation.

Some of the semi-technical descriptions of these twenty fundamental features have never before appeared in print.

The booklet was written by Chief Engineer A. C. Shaney, and is available to readers upon receipt of a 3¢ stamp to cover postage. Address: Amplifier Corp. of America 396-3 Broadway, New York 13, N. Y.



### SHURE CATALOGS

The Shure 1947-48 catalogs feature the new simplified and complete Shure line of microphones and pickups, a line designed to streamline inventory and build greater sales.

Catalog 157 illustrates the Shure Microphones, featuring the new Multi-Impedance "Unidyne" and "Sonodyne" Dynamics, the "Econodyne" Dynamics, and the two new crystal microphones, the "Monoplex" and the "Versatex."

Catalog 158 illustrates the Shure pickups, featuring the new "Muted Stylus" pickup, cartridge, and needles, the cartridge replacement "Pack," and the Shure lever-type cartridges.

# THIS MONTH

### FM FREQUENCY ALLOCATIONS

The proposal of the Federal Communications Commission to modify the allocation plan for F.M. station assignments, which in effect would double the frequency separation between adjacent stations in the same community, would make possible lower priced F.M. receivers than heretofore available, according to R. B. Dome, Electrical Consultant, Receiver Division, General Electric Company.

He said that field tests made under actual broadcasting conditions have indicated that doubling the frequency spacing from the present 400 kc to the proposed 800 kc would permit the reduction of the number of tuned circuits from eight, as needed at present, to about four.

### E-V APPOINTS BEIER

Electro-Voice, Inc., Buchanan, Michigan, manufacturers of microphones, floor stands, and other acoustic products, announces the appointment of LeRoy W. Beier Co., manufacturers' representatives, 600 S. Michigan Ave., Chicago, as sales representatives for the state of Wisconsin, eastern Iowa, and the northern two-thirds of the state of Illinois, excepting Chicago.

### WENDELL ELECTED

E. W. Wendell, Vice President in Charge of Federal Telephone and Radio Corporation, Clifton, New Jersey, American manufacturing affiliate of International Telephone and Telegraph Corporation, has been elected a director of the Radio Manufacturers' Association for a three year term.

Mr. Wendell joined the I. T. & T. system in 1925 and has been with Federal since the organization was formed He also is a Fellow in the Institute of Radio Engineers.

### NEW AMPLIFIER

Electronic Sound Engineering Co., 4344 W. Armitage Ave., Chicago 39, Ill., have announced that the first license for the manufacture of its new, high fidelity amplifier circuit has been granted to Universal Broadcast Equipment Corporation.

### MASCO ELECTRIC PHONOGRAPHS

Masco electric phonographs are among the most popular record players for home use, it was revealed by a recent survey made by Collier's magazine.

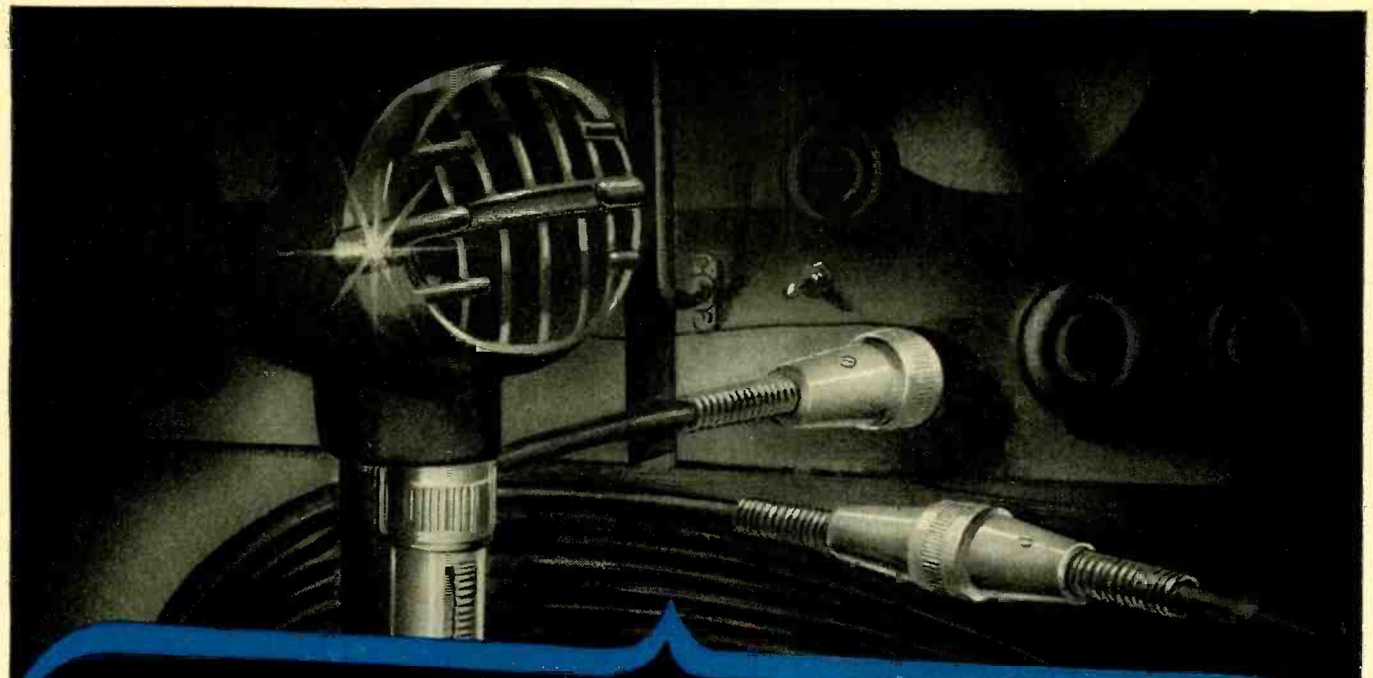
Made by the Mark Simpson Manufacturing Company, Inc., of Long Island City, Masco electric record players are but one of the items in the complete line of sound equipment and accessories manufactured by this pioneer in the sound equipment field.

# CORRECTIONS

Due to an oversight, the advertisement for the Amplifier Corp. of America, scheduled for our June issue, was omitted. It appears in this issue.

An unfortunate typographical error on page 8 of our June issue made the text read "maximum distortion and highest efficiency". It should have read "maximum power output with minimum distortion and highest efficiency".

### ADVERTISING INDEX



# LOWEST LOSS from Microphone to Chassis with Amphenol Connectors and Cables

Rated high in dielectric value, and with an improved power factor, Amphenol low-loss Microphone Connectors and Cables insure maximum efficiency in sound equipment. Dependable and easy to install, they are widely used by leading manufacturers of sound equipment apparatus, photo electric devices, home recorders and a complete range of similar items.

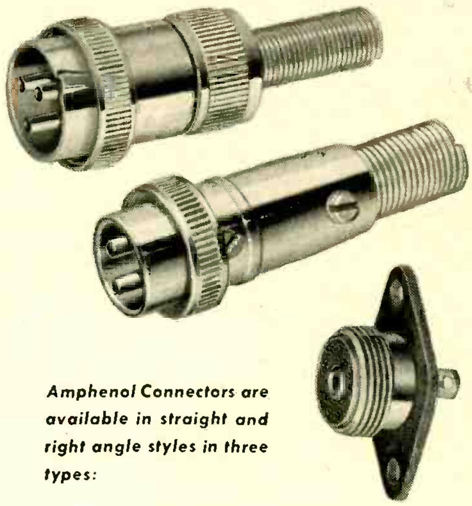
The Amphenol line is precision-built, compact, lightweight and complete. Microphone plugs, mating connectors and receptacles have unbreakable, chromium-plated brass shells and low-loss bakelite inserts. Cord protectors or cable clamps relieve strain on contacts. Threaded coupling rings screw onto mating coupling threads to provide "yank-proof" connections that are free from noise, leaks or shorts. Accidental disconnects are impossible. Multi-contact connectors are polarized.

**AMPHENOL MICROPHONE CABLES** are unusually low in capacitance for their small diameters.* Light in weight, their durable plastic jackets remain flexible down to —40° F. May be used with standard connectors and cord protectors.

Cables are available in either of two plastic jackets: Vinyl, for heavy auditorium and outdoor use where cables are long and the trampling of crowds is a factor; polyethylene for home and cocktail lounge applications.

Write today for new booklet describing Amphenol Microphone Connectors and Cables.

*Capacitance per foot ranges from 20 mmf. for Amphenol No. 21-120 (.242" diam.) to 35 mmf. for 21-156 (.155" diam.).*



*Amphenol Connectors are available in straight and right angle styles in three types:*

**SERIES 91:** Three and four contact microphone style, used on all types of portable apparatus. Since 1934, the standard of the sound industry.

**SERIES 80:** For standard single and two wire shielded cables such as: coaxial and microphone cables, twisted pairs, concentric lines, photo cell leads, etc.

**SERIES 75:** For standard single conductor shielded cable. Widely used in amplifiers, transmitters, photo electric devices, home recorders and similar equipment.

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*A Four-Way Performer*

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- A HIGH FIDELITY RECORD PLAYER
- A SENSITIVE, SELECTIVE RADIO RECEIVER
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Here is the combination you've been waiting for and the outstanding entertainment value of the year. A four-way performer, the new Meissner Radio-Phono-Recorder combines the functions of a versatile home recorder; a high quality, high-fidelity record player; a sensitive seven-tube super heterodyne radio receiver and a low power public address system . . . all in one unit that easily surpasses anything now being offered. Compact in design, precision-built and easy to operate, this super combination is now available at your jobber's. See it today or write for full information to the address below;

### THE MEISSNER RADIO-PHONO-RECORDER

**A Versatile Recorder.** For faithful reproduction of all radio programs or "live" talent. Crystal cutting head makes perfect records up to 10" in diameter. Crystal microphone. 2 turntable speeds for recording and playback.

**High Fidelity Record Player.** For playback of home recordings or commercial discs up to 12". Featherweight crystal pick-up arms. Ample storage space in lid for blanks.

**Sensitive Superheterodyne Receiver. 6 Tubes Plus Rectifier.** A sensitive, selective radio receiver built in for direct program recording without room noise. Both radio programs and "live" talent may be blended simultaneously on one record.

**Public Address Amplifier.** For "announcing" to small groups. Ideal for school & church use. May be connected to external speaker for greater volume.

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