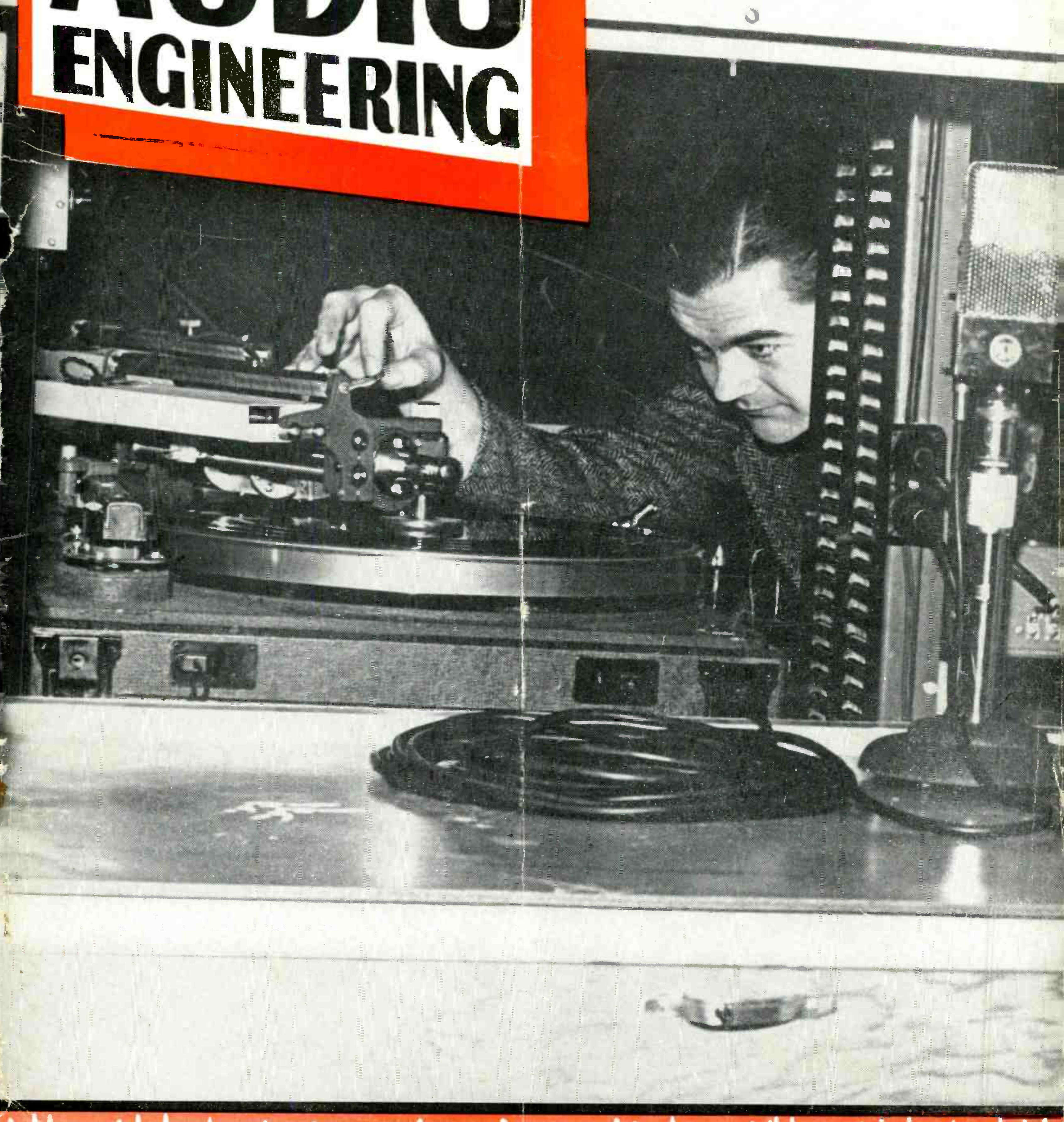
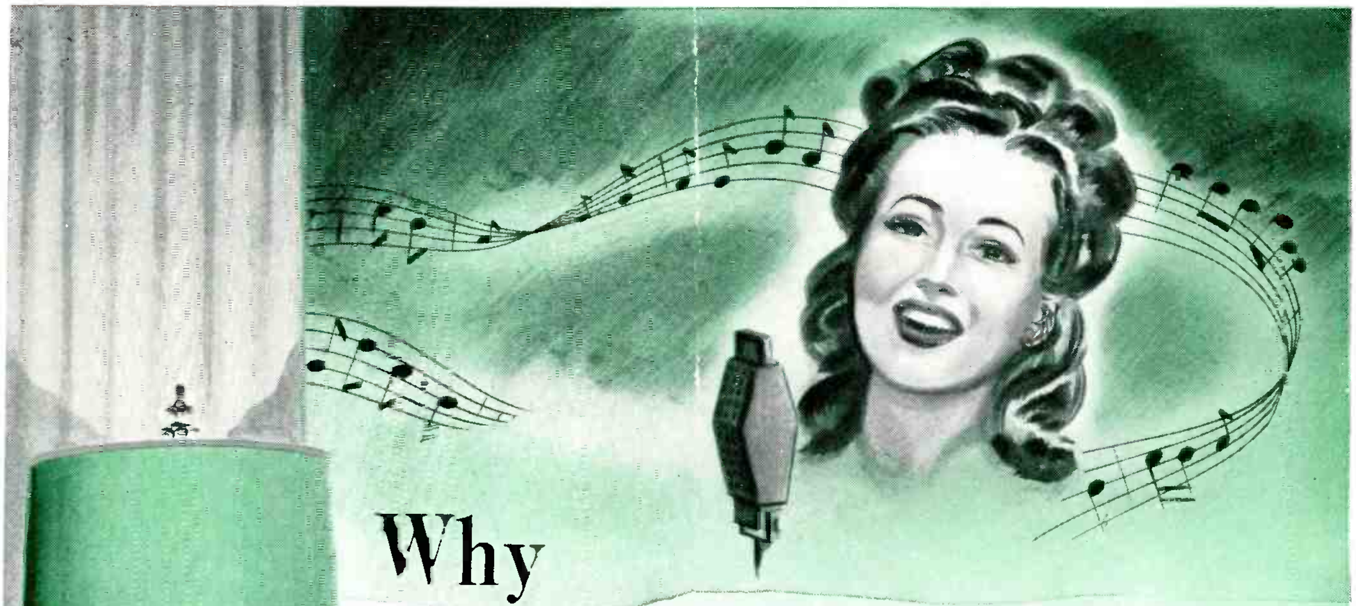
AUDIO ENGINEERING

SEPTEMBER
1947



THE JOURNAL FOR SOUND ENGINEERS



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COVER ILLUSTRATION

Rear view of Broadcast Recorder's new mobile broadcasting unit, showing recording machines, amplifiers, and table microphone in foreground. Chief engineer Joe E. Otis is using apparatus.

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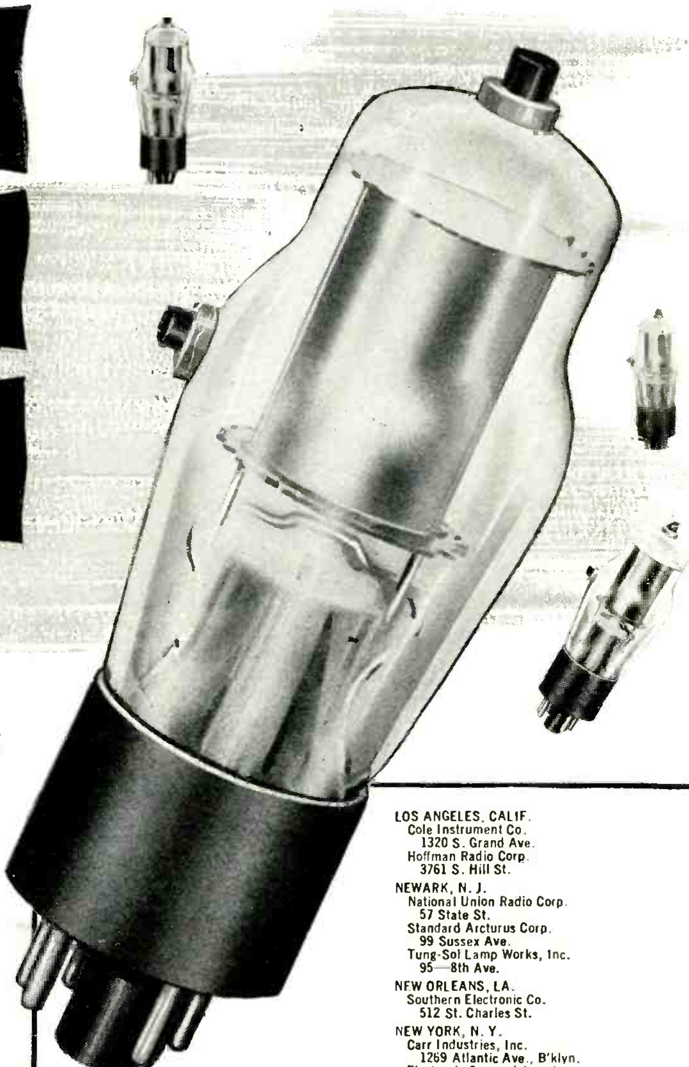
ELECTRONIC EQUIPMENT BARGAINS



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Letters

MORE ABOUT PITCH

Sir:

I should like to take exception to the viewpoint which you expressed in your editorial on Pitch, in the August issue of *Audio Engineering*—namely, that in the interest of achieving higher fidelity in reproduction, the concert pitch used during a composer's time be used today in performing his works.

First, a fundamental truth about music is that it is the inter-relationship of tones, rather than the individual tones themselves, which give "meaning" to music. Concert pitch is merely a technical convenience, useful as a mode of reference in any specific musical era, and established mainly by performers, conductors, musical instrument makers and physicists. The "content" of any musical composition is not affected by the relatively small historical variation in concert pitch; it is this "content" which the composer is anxious to convey to the listener.

Variations in concert pitch extend, historically, from A = 373.7 cps ("church pitch," 1648, ref. Mersenne) and A = 402.9 cps ("chamber pitch," same date and ref.) to our present A = 440 cps. The basis for the widespread variation is a complex one, involving such factors as hearing, room acoustics, scale temperament, the "temperaments" of performer and conductors, musical instrument construction and durability (e.g., church organs), style in musical composition, and virtuousity in performance. The attempt to standardize concert pitch at A = 440 cps (International Conference on Pitch, May 1939, London) was the inevitable conclusion of physicists and musicologists who were aware of the necessity for such standardization in modern society, whose technology makes cultural interchange both desirable and relatively easy.

Second, even in a specific era, different composers were accustomed to different, or similar, pitches, according to the instruments they used and the cities in which they happened to reside. Consider the travels of Mozart—France, England Italy, Germany; and consider that known concert pitch ranged from A = 395 cps, (organ at Trinity College, Cambridge, 1759) to Handel's tuning fork, A = 522.5 cps (London, 1751). How should one even begin to make a specification for the use of an "historical" concert pitch?

Third, many factors other than "historical" pitch are involved in any attempt to recreate the historical setting of a musical composition; so many, in fact, that such an attempt becomes more the work of a museum rather than a concert organization. For example, the violin of Bach's time was played with a bow whose wooden arch curved away from the bowhair; whereas the modern bow uses an inverted arch, permitting in-

[*Continued on page 5*]

EDITOR'S REPORT

JUST the other day, one of the most prominent broadcast engineers remarked that our magazine had brought to him a realization that there is a great deal more to audio engineering than he had formerly thought. This is quite understandable, because so little about the audio field is published elsewhere. The r-f men have been the glamor boys of the radio-electronic industry. It is their work which has received such disproportionate emphasis in both radio and general publications. Their accomplishments in making radio waves lie down, roll over, and bounce around have been oh'd and ah'd by the general public as well as the engineering professions.

But anyone who listens to an unmodulated radio wave gets awfully bored in a very short time. Nothing so forcefully points up the fact that an unmodulated wave is a carrier, and nothing more. Insofar as the broadcasting industry is concerned, the audio man has by far the toughest job. He must pick up sound, often under the most adverse circumstances, compress it within the dynamic range limitations of the r-f transmitter, keep distortion of all kinds to a negligible level, hold the esthetic qualities of symphonic music, and bear the criticism of tin-eared listeners using poor radio receivers. The audio man is the unsung hero of the broadcasting industry.

NATIONAL ELECTRONICS CONFERENCE

On my desk before me is a program of the forthcoming National Electronics Conference, to be held in Chicago, Nov. 3, 4, and 5 inclusive. This Conference is sponsored by the Illinois Institute of Technology, Northwestern University, the University of Illinois, the American Institute of Electrical Engineers, and the Institute of Radio Engineers—all in co-operation with the Chicago Technical Societies Council. Of the 74 technical papers scheduled for presentation, only two are within the audio engineering field. H. H. Scott will again discuss his dynamic noise suppressor, presenting, we understand, a more detailed description of this useful device. The only other paper is by Warren and Hewlett, and deals with intermodulation tests. Fortunately for those who want to get in and get out as quickly as possible, these two papers are the first to be presented. The program then veers off into a hodge-podge of instrumentation, microwave technique, nuclear physics, general electronics, and basic science, finally getting around to television and FM on the last day.

Most of these lectures are so highly specialized that those of interest to one group can have little appeal to others. If all discussions which deal with radio were subdivided into their proper categories and presented consecutively, the Conference would render a far greater service to those busy engineers who must travel appreciable distances to attend, and who have no time to waste. As it now stands, studies of nuclear reactions, the dynamic properties of the infra-red cesium arc, and scintillation counters, are sandwiched in between lectures on microwave technique, FM, and television broadcasting.

Just how the I. R. E. can justify its sponsorship of this program is difficult to understand. The Rochester Fall Meeting of the I. R. E., which follows right on the heels of this Chicago affair, is devoted entirely to radio and is always well attended. There is no earthly reason why Chicago shouldn't be able to offer a similar program—and audio, too, if you please—without including a dose of physics.

PROGRESS REPORT

Since our first issue in May, AUDIO ENGINEERING has enjoyed the most phenomenal growth in circulation of any journal in the radio-electronic field. A circulation audit during August showed that our paid circulation had nearly reached 10,000—almost double that of the first issue—and is growing at an even faster rate. Because we employ no subscription salesmen, circulation promotion concerns, or cut-rate offers to stimulate circulation, this growth must be due largely to the support given us by our readers. Let me take this opportunity to thank you one and all for your co-operation. It makes it easier for us to give you a better magazine.

To improve our service on the West Coast, Harry L. Bryant, Chief Engineer of Radio Recorders, will report on audio engineering developments in the Hollywood area. In October, Joseph P. Maxfield of our editorial advisory board will also establish himself in Hollywood as a consultant, and will contribute regularly to our magazine.

Forthcoming articles include one on the Magnetophon, studio model, about which there has heretofore been little written. A new principle in pick-up design is discussed by another writer. A complete series of p-a articles will start in the next issue.—*J. H. P.*

creased flexibility and virtuosity in performance, along with greater refinement in timbre. Would you, then, suggest that the interest of higher fidelity be served by requiring the use of the now-outmoded bow, in the performance of music by Bach and his contemporaries?

May I also mention the following objections:

Most instruments could be tuned to the required historical pitch; but what should be done with the piano, the organ, the xylophone, glockenspiel, etc.?

Today, the hearing phenomenon of absolute pitch is characterized by a generally reliable response to the reference pitch of A =440 cps—perhaps because of the conventional use of this frequency by piano manufacturers. The musician with absolute pitch (and there are many) hears and thinks of music in terms of the reference pitch (A =440 cps), whether he plays or composes. The mental gymnastics required of such musicians, if required to perform according to historical pitch, would be highly complicated, if possible at all.

I should be pleased to discuss this matter at greater length.

May I take this opportunity to congratulate you on your new magazine, and to express my hope that it will contribute to greater co-operation and understanding between the musician and the scientist? I should like especially to commend you on the series called "Musical Acoustics," by Mr. Tillson.

H. L. Robin
Director,
Acoustics Department,
Juillard School of Music
120 Claremont Ave.,
New York 27, N. Y.

GUIDING FISHES

Sir:

With regard to Mr. J. T. Barnaby's letter in your August issue. Answering his salmon problems, may I suggest that he look up "Forced Movements, Tropisms and Animal Conduct" by Jacques Loeb, M.D., Ph.D., Sc.D., (J. B. Lippencott Co., Philadelphia, 1918). In Chapter IV on "Galvanic trophisms," he shows that a large number of animals, such as shrimp, tadpoles, etc., when put in a trough of water through which a galvanic current flows, have a strong tendency to go to the anode, because they assume a position which causes them least pain. He also stresses the importance of showing the correct current density. In his experiments with shrimp he found that a current density of one milliampere per 1400 mm^2 rms best.

May I suggest that Mr. Barnaby try a direct electrical current through the water to guide his small salmon into safe channels past the Columbia River dams?

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

LISTENER PREFERENCE TESTS

Sir:

Dr. Olson's "Listener Preference Tests" seem to be by far the most valuable yet performed and apparently reverse the findings of the vast number of previous similar experiments. It seems to me that one important factor has been omitted in the reasons given for preference for a restricted range—high-frequency sounds are more attention arresting than those of lower frequency.

Prior to the advent of radio, listeners seldom heard music except when they made a considerable effort to do so, such as attending a concert. Under these circumstances, they attended to listen to the music, not to use it as a background for conversation or other activity.

With the advent of radio, music is now used chiefly as a background. People have developed an aversion to silence and often have the radio on from morning till night with only scattered periods of real listening.

Under these circumstances, the presence of high frequencies is attention compelling and therefore irritating, but if highs are cut heavily, a pleasant, unobtrusive rumble results which provides the required noise with minimum distraction.

In tests performed by Dr. Olson, the experience was presumably of short duration and the subject especially intended to listen to the music, which may account partially for the result.

I do not mean to imply in the foregoing that the other factors listed are not of

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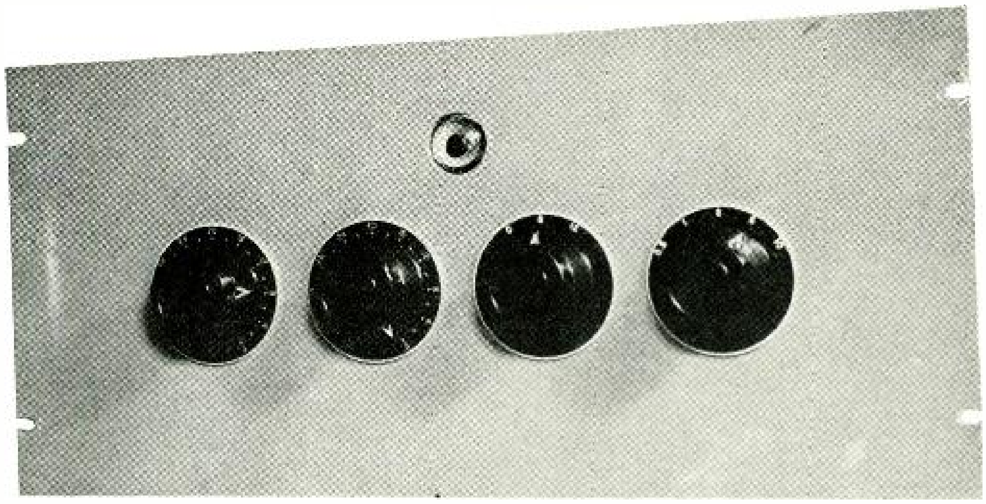


Front view of high-fidelity volume expander

High-Fidelity Volume Expander

NORMAN C. PICKERING*

Presenting for the first time a volume expander in which all forms of distortion are reduced to negligible proportions.

ALMOST ALL devoted listeners to recorded music, after they have gotten their playback systems to the point where the sound is pleasant to hear, feel that the loudness range on records is much too limited. To the listener who is concerned primarily with the aesthetics of the matter, the compromise can be made in either of two ways: the volume level can be set high enough to achieve satisfactory fortissimos—at which setting the pianissimos are too loud and too noisy; or the volume can be adjusted for maximum tolerable surface noise—in which case frustration sets in when the crescendos level off at a feeble mezzo-forte.

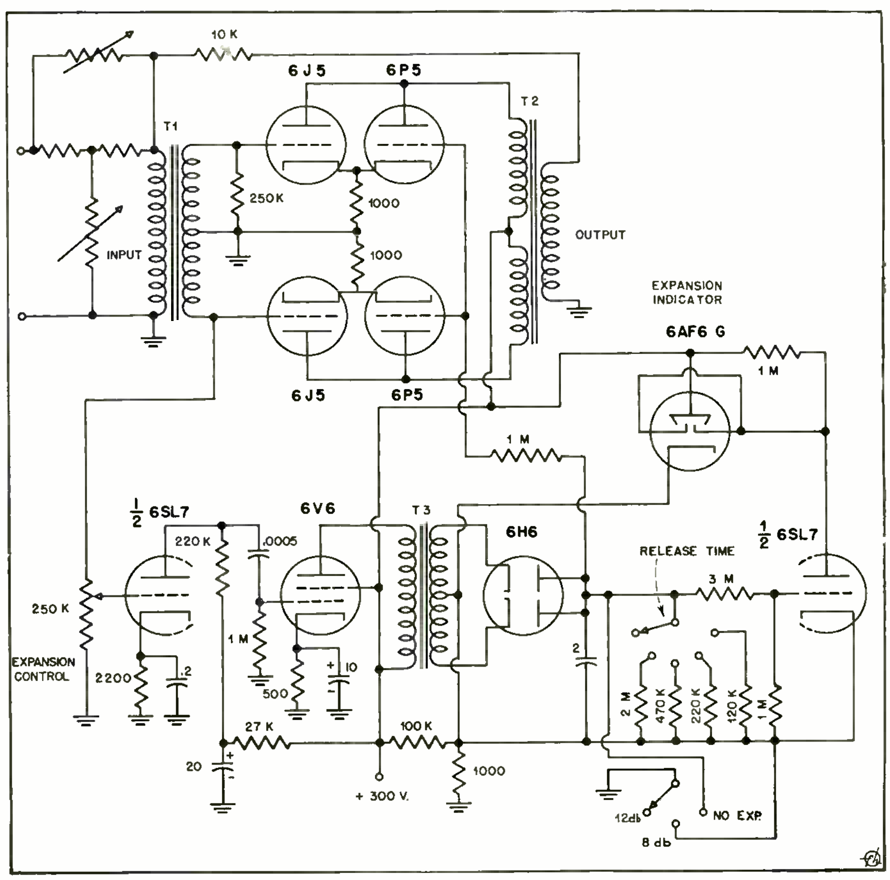
A considerable improvement in loudness range can be made by the intelligent use of a well-designed volume expander. The great difficulty is, of course, in the design of the expander itself, which is far from being a simple job. There have been many circuits offered in the past, all of which have at least one of the following defects:

1. The distortion is excessive
2. The expansion is not smooth
3. The rate of action is too slow
4. There is no positive limit to the amount of expansion.

The development of the present circuit was instigated by David Hall, the eminent authority on records and sound reproduction. The equipment had to pass the most stringent tests from the standpoints of both technical and musical listeners. This it seems to have done very satisfactorily, and it is believed that this is the most musically satisfying volume expander yet developed.

In order to keep the distortion to substantially zero level, and at the same time provide a definite limit to the possible increase in gain, it was decided to attack the problem by decreasing the gain of a high-quality low-distortion amplifier stage in a controlled manner, by a method which, in itself, would not introduce distortion.

Fig. 1. Schematic diagram of the high-fidelity volume expander.



*Pickering & Co., Inc.,
309 Woods Ave., Oceanside, N. Y.

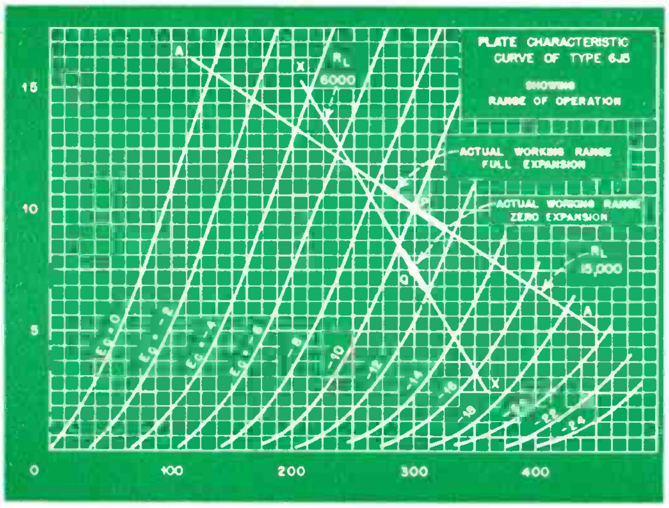


Fig. 2. Curves showing operating range of expander tubes.

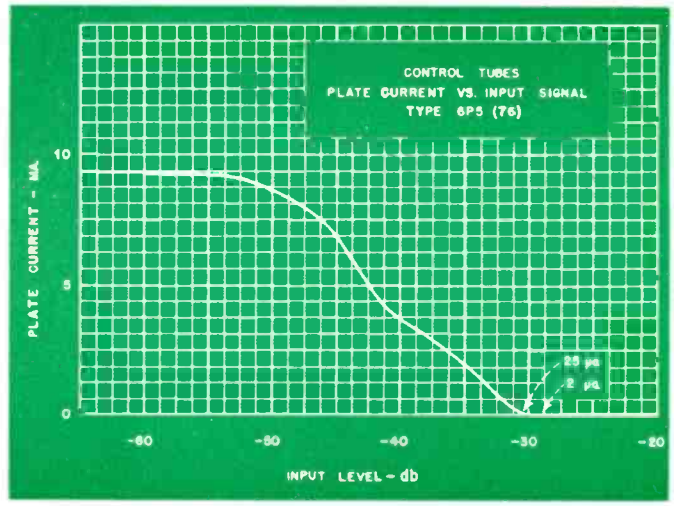


Fig. 3. Plate current vs. input signal curves of control tubes.

The use of variable-mu tubes was abandoned at the outset, as soon as inherent intermodulation distortion of the order of 15% was discovered. From the standpoint of low distortion there is still nothing better than a triode of low plate resistance.

The gain of an amplifier tube is given by the following equation*:

$$\textit{Voltage across load} = \frac{-\mu e_s Z_l}{R_p + Z_l}$$

where μ = amplification factor
e_s = input signal
Z_l = load impedance
R_p = plate resistance.

Since we have already ruled out the advisability of a variable-mu tube, we must find another way to vary the voltage across the load for a constant input signal, e_s. It is apparent that variations in the load impedance Z_l will change the output, especially if R_p can be varied in the opposite direction at the same time.

Referring to the circuit diagram of *Fig. 1*, it can be seen that the controlled amplifier is essentially a single stage of push-pull amplification. In addition to the plates of the amplifier tubes (6J5) the output transformer supplies current to the plates of the control tubes (6P5). The amount of plate current, and hence the plate resistance, of the control tubes can be determined easily by the instantaneous grid potential. At the same time, the additional plate current flowing through the common cathode resistor causes the plate resistance of the amplifier tubes to increase because of the increase in grid bias.

The actual operation is shown in the curves of *Fig. 2*. The output transformer is loaded with its nominal load of 500 ohms, which reflects a normal load of 15,000 ohms across each amplifier tube (control tubes cut off). This is indicated by the load line *A-A* across the curves. The quiescent operating point is shown by *P* in the diagram. The heavy line, of which *P* is the center, shows the region of actual operation. When a plate current of approximately 10 ma per tube is allowed to flow through the control tubes, their plate resistance drops to 10,000 ohms. This causes the effective load to be:

$$\frac{R_p \quad Z_l}{R_p + Z_l} = \frac{10{,}000 \times 15{,}000}{10{,}000 + 15{,}000} = 6000 \text{ ohms}$$

At the same time the effective increase in cathode current causes the operating point to drop to point *Q*. This condition of operation is indicated by the load line *X-X* passing through point *Q*.

The net result of the two effects is to reduce the amplification as the plate current of the control tubes is increased. The only thing which limits the amount of gain reduction possible is the approach of point *Q* to the non-linear region of the plate characteristic, with resulting increase of intermodulation distortion. It is interesting to note that the limitation in permissible distortion is on the low-level or unexpanded condition of operation. There is no possibility of running into distortion when the amplifier is wide open.

With the control tubes cut off, the gain of the amplifier stage is approximately 20 db. This will be the maximum possible gain, regardless of the amount of expansion used. A large number of listening tests indicated that, if smoothly applied, a maximum of 12 db expansion is permissible on well-monitored records. On records where there are noticeable "steps" in the volume changes due to too-rapid changes of gain during recording, 8 db is the maximum tolerable amount of expansion. All of these results were based on the premise that the listener must not be able to detect the actual operation of the volume expander.

With the foregoing information as a basis, the type of control tube was chosen, and the operating point was selected to give a loss in gain of approximately 8 db when the control grids were at zero potential relative to ground. It was then found that a grid voltage of –21 volts was sufficient to reduce the plate current to 2 microamperes. A positive potential (relative to ground) of 3 volts, applied to the

*Terman—Radio Engineering, p 173.

Fig. 4. Input signal level vs. expansion curves.

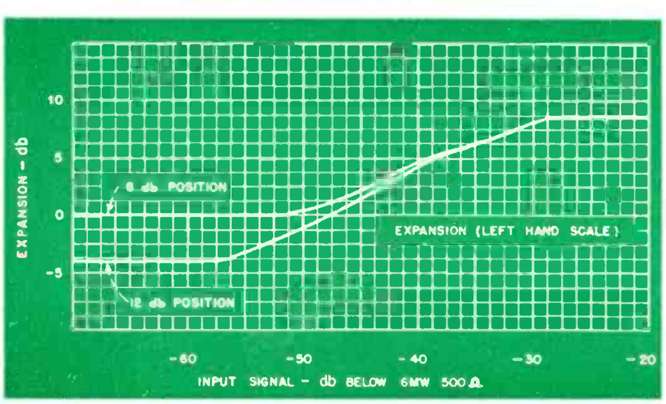
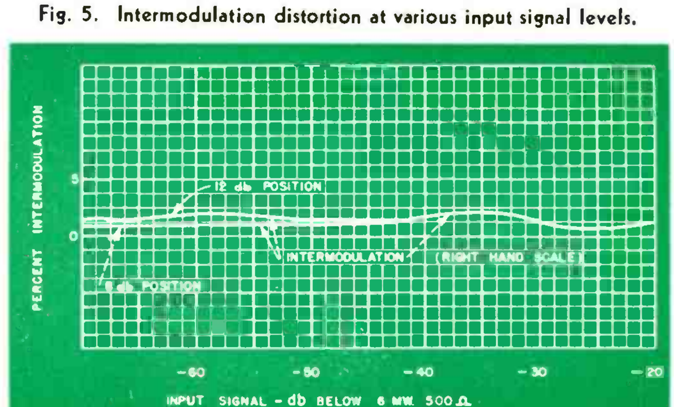


Fig. 5. Intermodulation distortion at various input signal levels.



control grids, decreased the gain by an additional 4 db. It is necessary merely to switch this positive bias on and off in order to change the expansion range from 8 db to 12 db. An additional switch position is provided to ground the control grids, making the unit a fixed-gain amplifier with zero expansion.

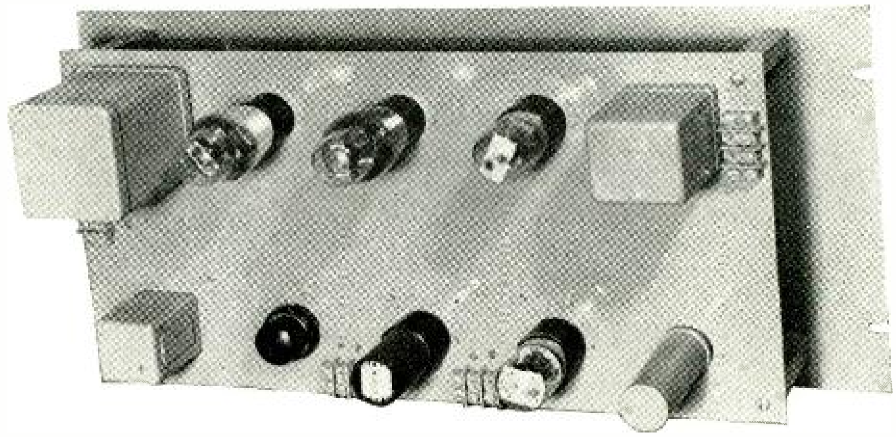
Time Constants

The next problem is that of the attack and release time constants of the system. Again it was necessary to resort to listening tests to decide on the optimum conditions. It was quickly decided that an instantaneous attack was necessary to provide unnoticeable operation. In other words, a loud sustained tone reaches its full level at once, and doesn't build up over a noticeable time interval, as so often happens with volume expanders. On the other hand, it is necessary to delay the release time so that the expander will not follow individual cycles at low frequencies. Furthermore, the release time adds to the normal decay of the sound, and, if too rapid, will effectively decrease the amount of reverberation. For this reason it is necessary to have at least 2 seconds delay for normal program material. In the laboratory model of the expander, values of 0.5, 1, 2, 5 and 10 seconds were provided. It was found that 5 seconds gave good results on almost every type of record.

Inasmuch as it is necessary to charge the 2 mfd. capacitor in a matter of 1 millisecond or so, it is apparent that a considerable amount of power is required. A 6V6 is used to supply power to the rectifier. The rectifier, type 6H6, is momentarily called upon to supply several times its peak rated current of 10 ma, but because of the extremely short duty cycle, this has no adverse effect on the tube life.

It was necessary to do a considerable amount of testing to determine the optimum frequency response of the amplifier used to supply power to the diode. If a wide-band—30-10,000 cps—frequency range is used, it is apparent that the control voltage will be influenced by the strongest signal currents within that frequency range. Furthermore, low-frequency rumble and thumps, and high-frequency surface noise and clicks, will produce false control and consequent gain increase. It is best, therefore, when using a single band of frequencies from which to derive the control signal, to limit the response to from 500 to 3000 cycles. The effective loudness is determined by this band of frequencies, anyhow, so it is logical to use the same region for the controlling voltage.

An interesting effect was achieved by splitting the control amplifier into two channels, each supplying a separate diode. The outputs of the diodes were connected in series, and the individual amplifiers were arranged so that they would overload when supplying half the necessary control voltage. One amplifier passed the frequency range from 100 to 500 cycles; the other from 600 to 3000 cycles. Either amplifier alone could only produce half the total expansion, regardless of the energy in the pass band. It required energy in both bands to produce full expansion. This condition prevails in full orchestra, organ, or band music, at which time full expansion is required. This system prevented "blasting" when a single instrument or voice momentarily overbalanced the full ensemble. The results were very good, but on most records the improvement over the single-channel system of restricted frequency range did not justify the circuit complexity.



Rear view of high-fidelity volume expander.

Expansion Indicator

A 6AF6 electron-ray tube is used as an indicator of the amount of expansion. It has a considerable advantage over a pointer-type meter, because of its freedom from dynamic error. It is adjusted so that the eye is just closed when the 6P5 control tubes are cut off.

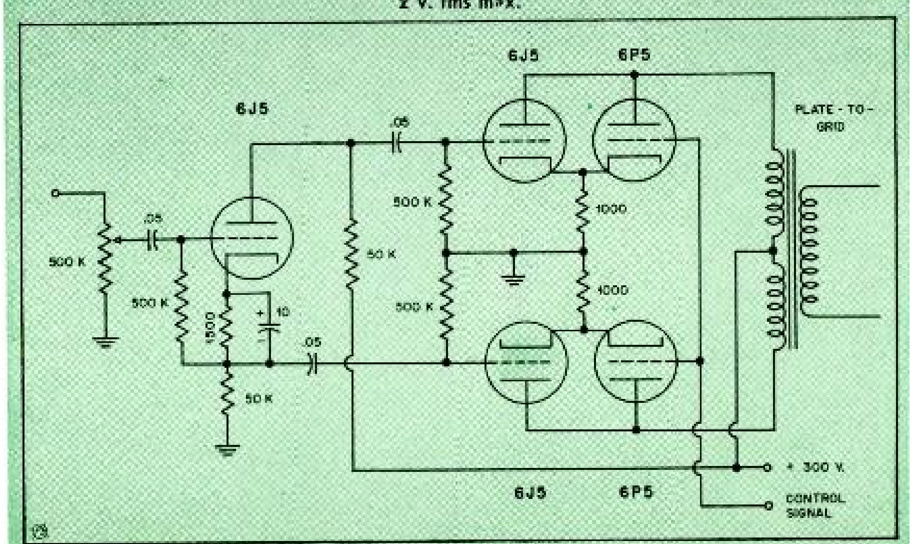
The last important problem is the linearity of expansion versus input signal. It is very important not to have any "steps" in the expansion control. The plate-current variations with input signal, with the control voltage derived from the previously described rectifier system, are shown in *Fig. 3*. The change in plate resistance is not exactly the type of curve needed to produce a linear expansion. This condition was greatly improved by making the total range of expansion about 6 db more than desired, and reducing the over-all gain by means of inverse feedback. Since the gain reduction is a function of the amplifier gain, it is apparent that the amount of feedback will vary as the amplifier gain is varied. The gain will be reduced more at full amplification than at low levels. This has the effect of straightening out the expansion curve and reducing its slope, thereby accomplishing the desired end. This is shown in *Fig. 4*. The feedback also contributes somewhat to the over-all low distortion obtained with this device. The curves of intermodulation distortion versus input signal level are shown in *Fig. 5*. The harmonic distortion is barely measurable.

Fig. 6 shows the main amplifier circuit arranged to be inserted in a high-impedance amplifier. The action is identical with that of the low-impedance unit.

It will be noticed that this circuit can be used as a remotely operated gain con-

[*Continued on page 39*]

Fig. 6. Modifications needed for high-impedance input circuit. Input signal should be about 2 v. rms max.



Multi-Lingual Interpreting Systems

C. A. TUTHILL*

Describing a unique and most useful application of audio equipment.

EXCEPT IN OUR STATE DEPARTMENT, little is known in this country about simultaneous translation from one into several languages. In Europe two adjoining states—such as Pennsylvania and New Jersey here—speak two different tongues. To countries on the continent it is an old story, but until the United Nations invaded these shores little concern was ever had hereabouts for complex translation.

Until audio interpreting systems were installed in three conference rooms last fall at Lake Success, L. I., many thousands of man-hours were lost during United Nations debates. More than one hundred fifty delegates and their aides had to sit restlessly through draggy, annoying interpretations of a speech. The meetings would come to a complete halt while trained interpreters repeated in French or English the twenty-minute or half-hour speeches. The Russians generally spoke in their own language for home shortwave consumption. That called for two translations. That caused the clock hands to travel three times as far as necessary with simultaneous facilities.

This nuisance was partially eliminated one year ago when the *electronic* was coupled to the *human* with such success that five major languages were poured back into the ears of the delegates as rapidly as their colleagues spoke. As soon as a speaker concluded, the other delegates knew what he had said. Today the representative from China watches his friend from Paris orate in French while he hears the speech in his own Chinese. Larger conferences and committees now schedule their meetings so as to obtain a room fitted with multi-lingual facilities rather than endure the imperative delay of verbal repetition in rooms not yet equipped.

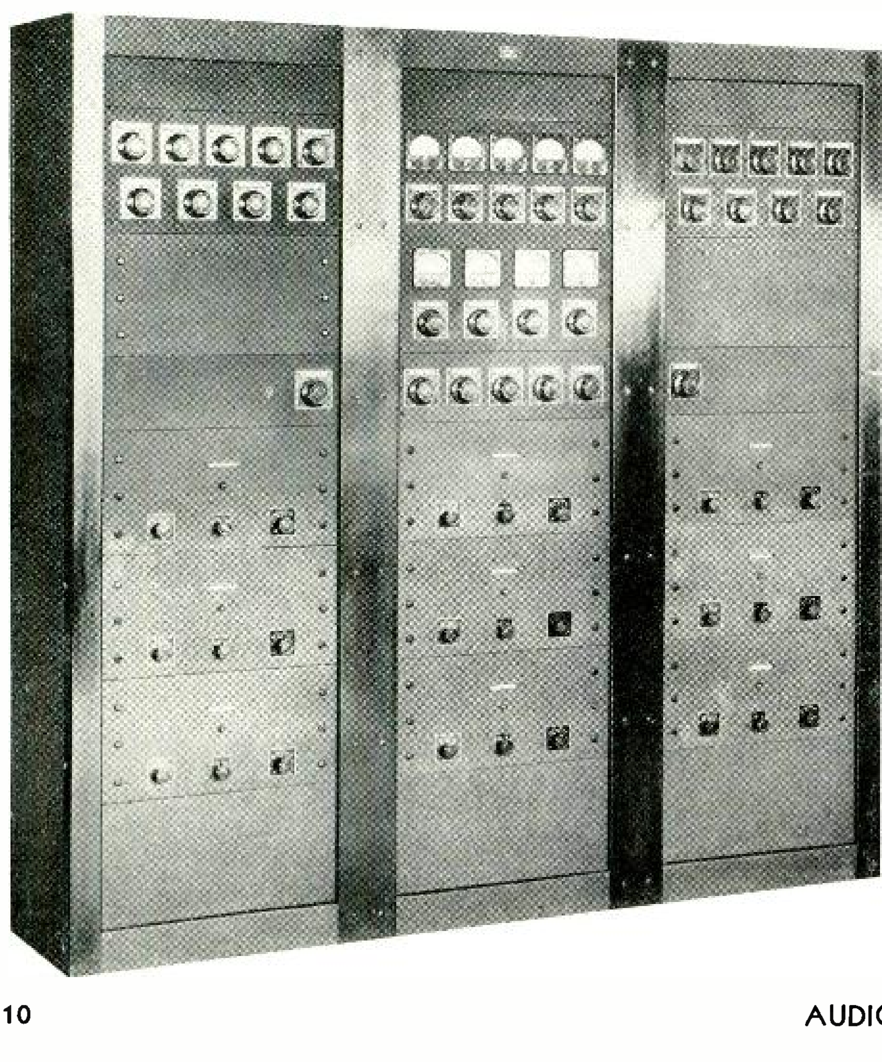
This great time-saver proved its worth more than fifteen years ago in international conferences. After the old League of Nations settled in Geneva, nine IBM-Filene-Finlay Translator audio channels were installed there. More recently the prosecutors of the Japanese and German War Trials have used this same equipment. It has been roughly estimated by one of the judges participating at the Nuremberg trials that those hearings would have consumed five or more grueling years were it not for the multi-lingual channels.

*2345 Broadway, New York 24, N. Y.

Conference Rooms

At United Nations headquarters, two large conference rooms served by the multi-lingual equipment seat over fifty delegates each around an oval-shaped table. Selector switches and a headphone to the right of each national offers him a listening choice of five languages. Any delegate understanding the spoken tongue need wear no headphone, since both the conference table and outer areas of the room are covered by a low-level p-a system utilizing 48 speakers, with those adjacent to microphones being killed automatically by relays when the microphones are keyed into the circuit. Outside the inner circle the advisers, press, radio and any public overflow are also serviced. Selector switches and phones are mounted to the sides of chairs strapped mechanically together in banks. This construction protects cable feeders and expedites trouble-shooting through bank isolation.

Across the end of the room opposite the chairman, and elevated enough for good vision, are five sound-proof, air-conditioned booths. Each seats three interpreters who relieve each other in the handling of one language since there can be no interruption of service for human failure. In rotation laterally these booths handle: Chinese, Spanish, Russian, French and English. Adjacent to these is an open booth housing rack-mounted



Original IBM nine-channel rig used at League of Nations, Geneva, Switzerland. (IBM Filene-Finlay Translator.)

variable-gain amplifiers, rectifiers, patch panels and spare equipment. An engineer checks the entire system well in advance of operation and constantly monitors the channels at this point.

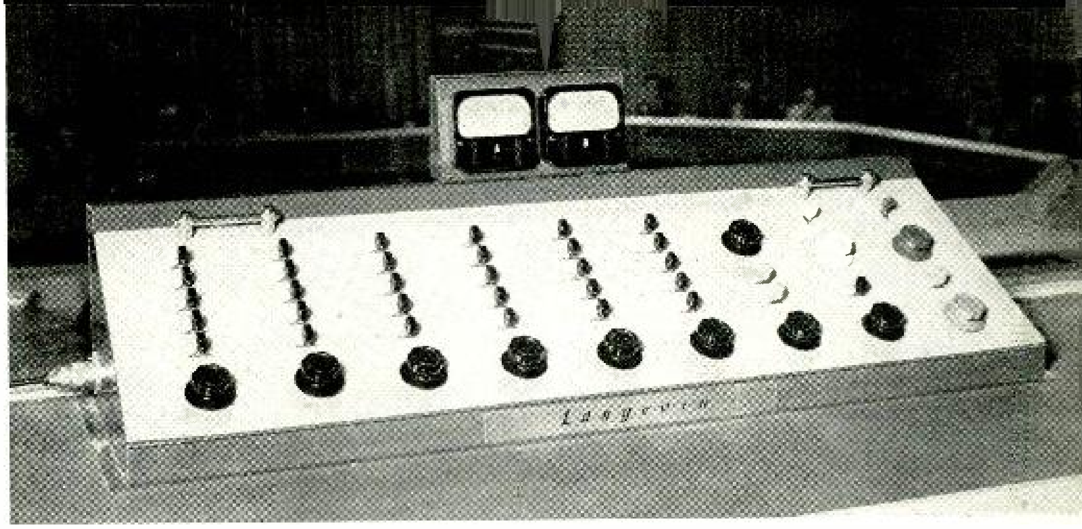


Fig. 2. UN Lake Success control unit. *(Courtesy The Langevin Co.)*

System Operation

Delegates in all major UN meeting rooms have microphones before them which normally feed all radio, television, newsreel recording, and p-a apparatus. It is from these microphones that the interpreting channels start to function (see block diagram, *Fig. 1*). From pre-amps on through a keying mixer and its boosters a feed is linked with the multi-lingual input. Here it is balanced, amplified, monitored and distributed to the headphones of fifteen highly capable male and female interpreters.

The keying-mixer, *Fig. 2*, is the audio nerve center of the entire system. The two vu meters shown centrally above the mixer panel indicate the behavior of the basic regular and emergency channels. Starting from the left of the panel, six vertical banks of five keys each cut on or off thirty microphones. These accommodate sixty delegates, each microphone serving two people. The knobs directly beneath the rows of keys pre-set and control levels. The two keys above the seventh knob from the left are for two interpreters' microphones when the room is not equipped for simultaneous interpretation. The knob above these keys controls a two-position, low-roll-off speech equalizer. The eighth knob and its microphone key allows individual handling of the chairman. The notable feature here is that the gain controls for thirty-three microphones are boiled down to eight knobs. The white knob and key above set the monitoring level, and provide instant comparison if either bus behavior is in question. The lower right knob is the final master. Its key instantaneously flips the output to an alternate line amplifier in case of trouble. Above the master is the p-a control.

A strategic factor in this arrangement is key designation. The engineer must find the correct key for an active nation in a split second. This calls for distinct labelling of all keys—two nations per key. For that reason a light-colored main panel is topped by a translucent plastic mat. This permits rapid markings and erasure. Full committees of fifty or more nations are often replaced by a similar one in the same room within fifteen minutes during busy days.

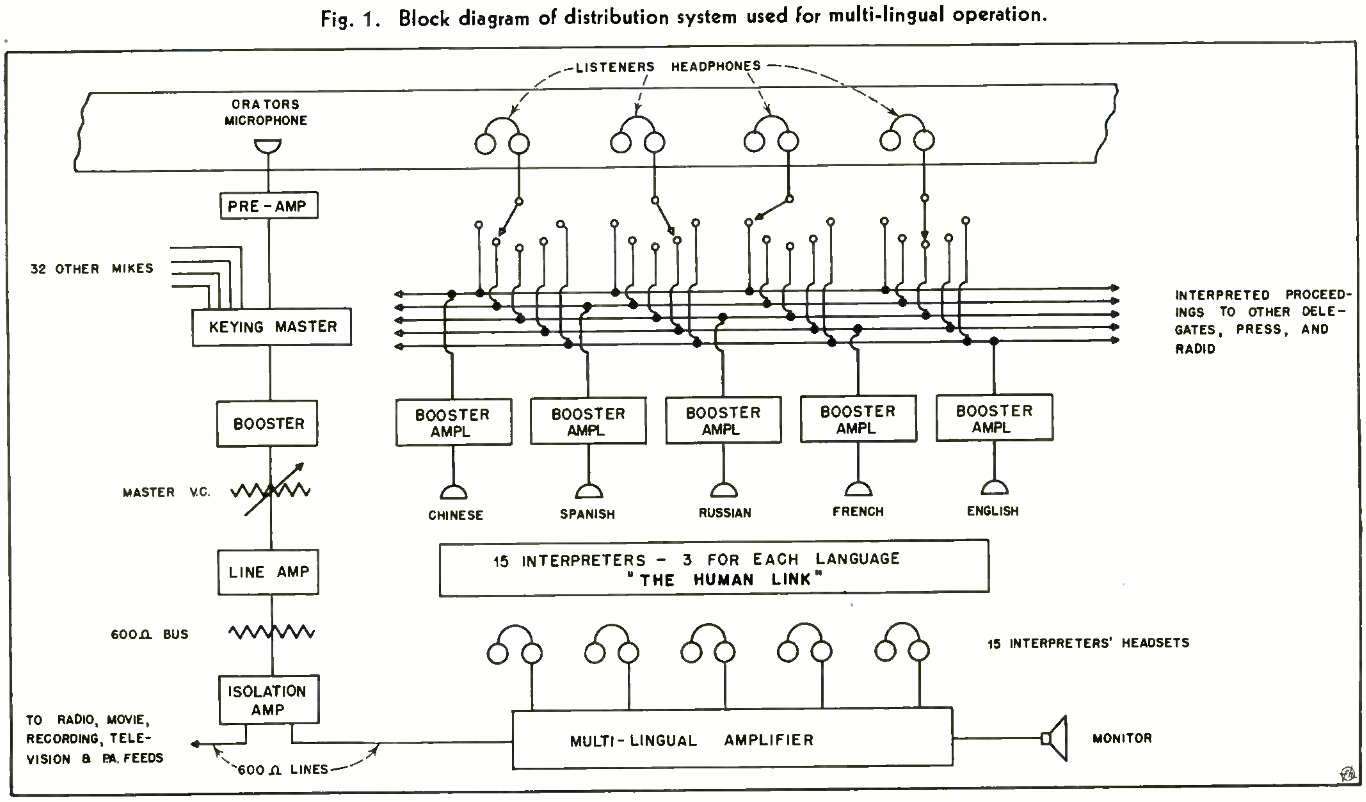
The alertness and good judgment of the control engineer contribute greatly to the smoothness of the show. The chairman frequently fails to recognize delegates and they often do not wait for recognition. They do not stand in conference while speaking. Since some are seated with their backs to the engineer, and are often entirely inanimate, he must recognize their voices to cut in the proper microphone. When argument reaches heated cross-fire proportions, he has difficulties indeed.

Interpreters

Referring to the block diagram, *Fig. 1*, we see the *human* factor linking the translating audio equipment with the original. As in the performance of preceding units along the line, each is dependent upon the other. Thus, at a second point, the entire system depends upon the human element.

The men and women who translate these international orations perform with the precision of pulsing circuits. Justice to their achievement cannot be done here. Their concentration and patience is laudable; many are tested but few are chosen. To relay truly the original speech, they must introduce no feelings of their own as they translate rapidly. Instead they must convey the genuine intent

Fig. 1. Block diagram of distribution system used for multi-lingual operation.



supported by enough bona-fide expression to hold attention.

Literal word-for-word interpretation being an impossibility on account of idioms, these multi-tongued people lean upon their own judgment for an equivalent phrase which will convey the identical meaning in another language. Despite all this, these human time-delay relays lag but slightly. They often finish within a second or two of the original speaker. These interpreters do an admirable job.

The audio channel picks up again as the interpreters talk into breast-type ribbon microphones. From this point forward the speech is boosted conventionally through variable-gain amplifiers constantly monitored. From the rack-mounted regular and spare amplifiers the interpretations are distributed over audio lines to several hundred selector switches and their headphones. With all seats in the house covered, the audio system is completed.

Radio System

An innovation of the above method deserves some treatment here even though it is intended to employ audio systems at the permanent UN headquarters in New York. This is a small radio installation, as shown in *Fig. 3*.

Fig. 4. IBM Filene-Finlay delegate desk mike.

In several IBM-Filene-Finlay installations and in the one which has served the World Radio Conference at Atlantic City, N. J., low-powered frequency-modulation transmitters have been installed. In the above case ten crystal-controlled low-frequency FM transmitters, all coupled to one loop antenna encircling the auditorium, are operated with carriers spaced twelve and one-half kilocycles apart. The channels run from 100 kc upward and provide adequate field strength although operated at low power.

In this installation a great amount of audio wiring is avoided. Only the microphones and signal lights need be wired. Instead, 1800 FM receivers weighing only one and one-half pounds each are provided for the delegates. They resemble and are worn like a flat camera, the shoulder straps doubling as loop antennas. The receiver dimensions are actually 1⅝ x 4⅛ x 5½ in. Not necessarily worn, they may even be buried under the delegate's papers while he works at his post. Each set has an off-and-on switch, a selector dial for the desired language and a volume control Each employs three midget tubes and affords two stages of radio amplification. a-v-c diode detector, and a pentode output stage permanently connected to headphones. Internal high-capacity war-developed batteries will handle one week of daily operation without replacement.

Delegates enjoy the freedom of being able to move about the hall, comparing notes with colleagues while still hearing speeches interpreted. When they want to talk, however, they must return to their seats and use a desk-type microphone,

[*Continued on page 46*]

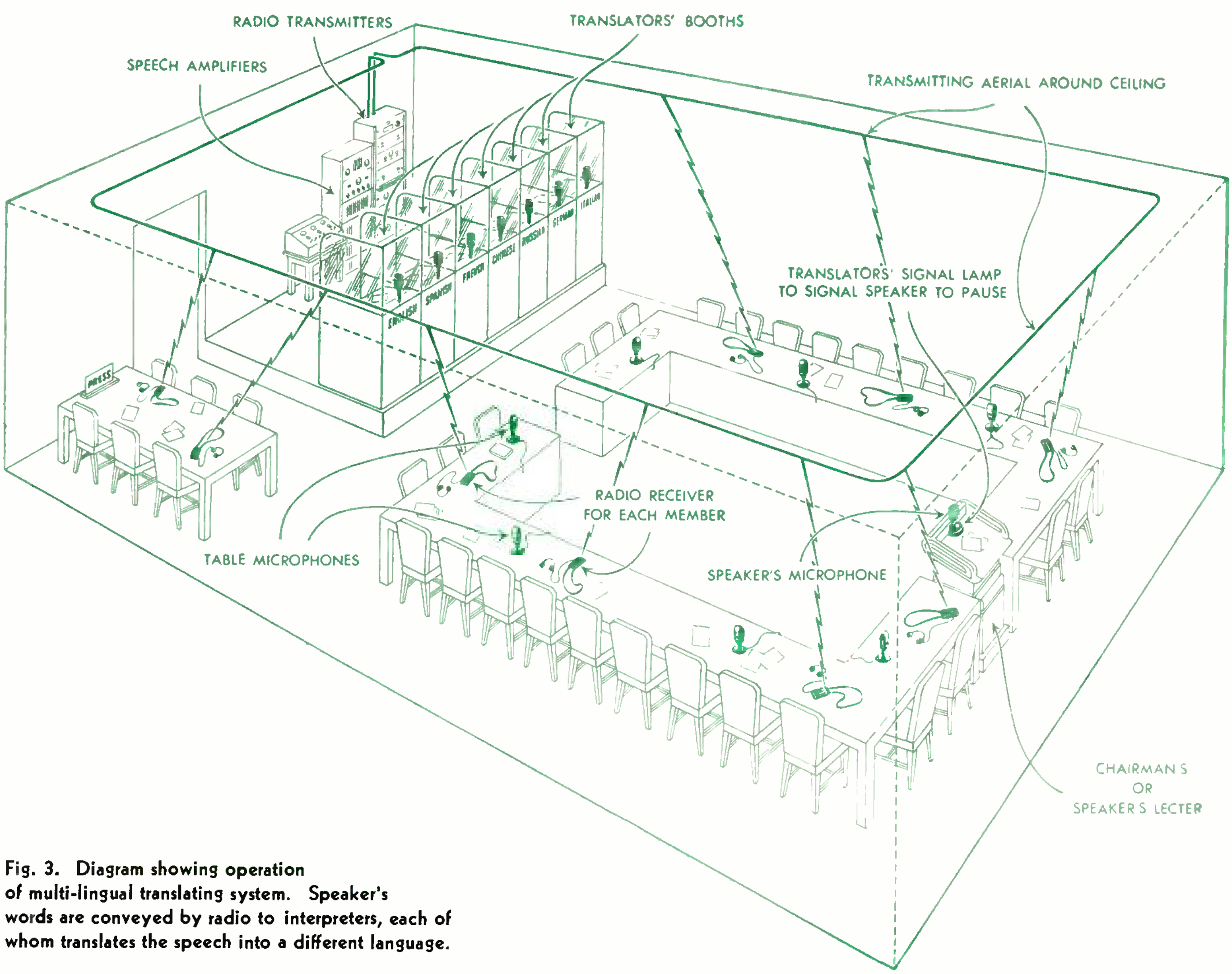


Fig. 3. Diagram showing operation of multi-lingual translating system. Speaker's words are conveyed by radio to interpreters, each of whom translates the speech into a different language.

High-Frequency Equalization for Magnetic Pickups

C. G. McPROUD*

Methods of using shunt capacitance to provide low-pass filter action and NAB compensation for users of Pickering and G-E pickups.

ONCE THE OUTPUT of a magnetic pickup has been equalized so that the response up to the transition frequency is a realistic reproduction of the original recorded material, the listener next turns to the other end of the spectrum in an attempt to eliminate noise or to match the recording characteristic. The methods of equalizing the low-frequency response have been covered previously†, and such methods are equally applicable to both the General Electric Variable Reluctance pickup or to the Pickering cartridge. These pickups are of the high-impedance type, designed to work directly into the grid of a vacuum tube. The electrical characteristics are similar, and the same methods of high-frequency equalization can be applied to either pickup.

In order of importance, the three principal reasons for equalizing the high-frequency end of the audio band are:

1. To reduce surface noise
2. To eliminate distortion
3. To compensate for the recording characteristic

The reasons for this order will be apparent from the discussion.

The first of these listings is the one most noticed by the layman, who is often the person for whom the equipment is being designed. He is apt to be conscious of noise much before his notice of distortion, and favorite records—usually played more than others—are likely to be the most worn and consequently the greatest offenders.

The second reason becomes more apparent as the ear acquires training. The effect is noticed on many records as a definite "muddiness" and "smearing" in the high-level, full-orchestra passages, probably as a result of cutting at an average level which is too high for the amplifier equipment, the cutting head, or the record material. Consequently, when peak passages are reached, system distortion rises considerably. A higher over-all distortion *which remains constant* is less objectionable than low distortion throughout parts of the recording, with great distortion on high-level passages.

Compensation for the recording characteristic is likely to be considered by many as the most important of the three listed reasons. Without diverting into a discussion of recording characteristics—which are definitely not standardized among the various manufacturers, or even in one manufacturer's own studios—let it be said that some degree of high-frequency equalization is generally employed in the recording studio, as measured from the microphone to cutting head. However, this does not necessarily mean that the individual record that you buy is going to reproduce with that increased h-f response. It is recognized that one stamper is capable of producing about 1,000 records before it is discarded. The first record may be excellent, but what about the 999th? One more record and the stamper will be discarded. In the writer's opinion, very few commercial records are on the market today with any excess of high-frequency response—certainly not to the extent that they cannot be compensated for easily by usual "tone control" circuits. Therefore, with the exception of compensation for the NAB characteristic, no "sloping" circuits will be discussed. Reproduction from the record will be assumed to be correct as far as frequency response is concerned, and since both of the high-quality magnetic pickups in growing use are capable of performance up to 10,000 cps, it will be further assumed that the electrical output from the pickup is directly proportional to the velocity of the stylus tip.

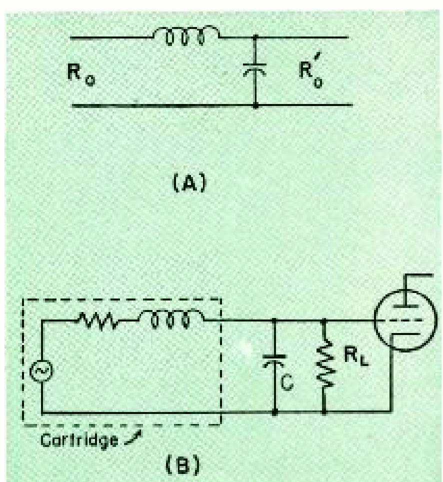


Fig. 1. (A) Conventional L-section low-pass filter configuration. (B) Circuit of magnetic pickup shunted by capacitor C and terminated by resistor R_L, feeding grid of amplifier tube.

Correction Methods

The simplest method of reducing the response of any electrical circuit to a desired maximum frequency is by means of a low-pass filter. In proper designs, practically any shape of curve can be obtained. However, in high-impedance circuits the series inductances reach large proportions, and they do pick up hum. Consider a simple low-pass filter such as that shown in *Fig. 1A*. It consists of a series inductance and a shunt capacitor. The equivalent circuit of a magnetic pickup is essentially a generator in series with an inductance equal to that of the coil, and with a resistance equal to the resistance of the coil. Thus, if a capacitor is shunted across the output of the pickup, as in *Fig. 1B*, the net result is that a low-pass filter section is formed. It is only necessary to choose the correct value for the capacitor and the correct value for the terminating resistance, after the cut-off frequency is selected.

L-P Filter Circuits

The configuration for the low-pass filter used in connection with these magnetic pickups is now seen to consist of the inductance of the pickup itself shunted by a capacitor, with the combination working into a suitable load resistance. Without considering the actual cut-off frequencies, the general circuit is developed first from the formulas for a constant-k low-pass filter, which will provide the values for C when the inductance L and the cut-off frequency are known. The cut-off frequency, f_c, is that at which the response is down 3 db.

* Managing Editor, *Audio Engineering*
† *Audio Engineering, p. 10,* July, 1947

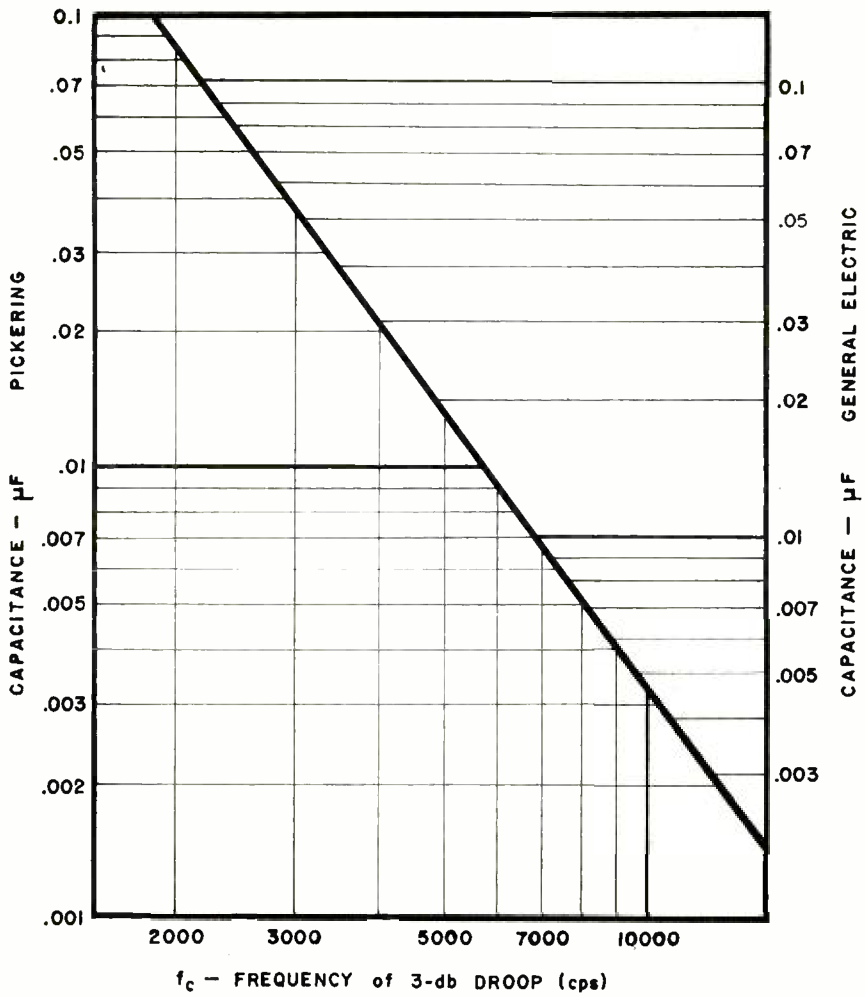


Fig. 2. Chart for determining capacitance to be shunted across Pickering Cartridge or General-Electric Variable Reluctance Pickup to provide low-pass filter action.

By further rearrangement of the formulas, the value for the terminating resistance is given in terms of L and C. To simplify the determination of these capacitance values, they are shown in chart form in *Fig. 2*. The inductance of the Pickering cartridge is approximately 120 mh, while that of the G-E Pickup is approximately 100 mh. To use the chart, determine the frequency at which a 3-db droop is desired; follow the frequency line upward until the diagonal line is reached. For the Pickering, read the correct capacitance value from the left edge of the chart; for the G-E, read the values at the right.

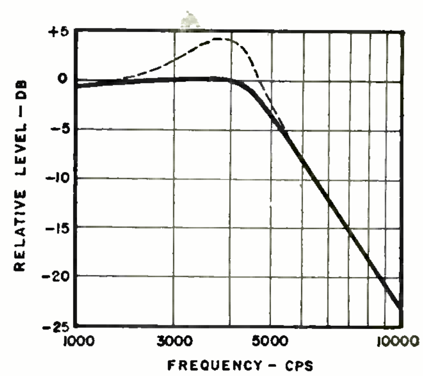


Fig. 3. Shape of response curve for $f_c = 3850$ cps. Dotted curve indicates peak due to improper terminating resistance.

Response Curves

A typical response curve obtained by the use of a properly terminated capacitance shunt across the output of a magnetic pickup is shown by the solid curve of *Fig. 3*, while the dotted curve shows the effect of operating the combination into a load resistance equal to three times the optimum value. The smoother response of the solid curve is much to be preferred, and the use of a filter with sharper cut-off characteristics is generally undesirable for high quality audio circuits. Empirical values for the terminating resistance are given by the simple relations

R (ohms) $= 1.2\,f_c$ for the Pickering, and

R (ohms) $= 0.9\,f_c$ for the General Electric.

The response is flat up to approximately 80 per cent of the cut-off frequency, is down 3 db at f_c, and decreases at the rate of approximately 15 db per octave above f_c.

Any resistance in series with the capacitor does not affect the curve up to a frequency of approximately $2f_c$. Above this point, the curve tends to flatten off. Therefore it is recommended that the capacitor be connected directly across the pickup, and shunted by the correct load resistor. For flexibility, and to provide for wide-open response when record surfaces and distortion content permit, a switch may be arranged to connect either of two or more capacitors across the circuit as desired, each capacitor having its own resistor permanently connected across it. Using this arrangement, cut-off frequencies anywhere in the band can be chosen at will. For ordinary use on commercial shellac pressings, a three-position switch providing cut-off frequencies of 3,500 and 6,000 cps, together with one wide-open position, has proven quite satisfactory, although a more elaborate arrangement would obviously give a wider range of control.

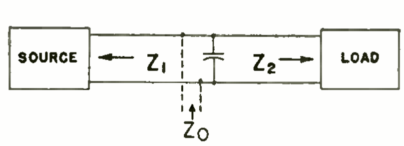


Fig. 5. Impedance Z_0 of any point in a circuit is composed of Z_1 and Z_2 in parallel.

NAB Roll-off

Professional users of these pickups often have need for a circuit which will provide the proper roll-off for the high frequencies to permit satisfactory reproduction of transcriptions employing the NAB recording characteristic. This curve is shown by the solid heavy line of *Fig. 4*, together with the equalization provided by the low-frequency equalizing circuits previously described, and the simple capacitance droop to correct for the high end. The resulting range of control provides for a response at 100 cps anywhere between +6 and −2db, together with an approximately flat response above 1,000 cps.

The droop resulting from the shunting of a capacitor across a line of a certain impedance is a gradual deviation from flat to a loss of 3 db at approximately three times the frequency at the point of deviation, with a further droop at a rate of 6 db per octave above that point. However, this does not apply to capacitor shunted directly across an inductive source such as a pickup. The capacitor must be shunted across a circuit which is essentially resistive. A simple method of determining the correct capacitance to provide equalization for the NAB curve is given by the following:

$$C = \frac{75\,(Z_1 + Z_2)}{Z_1 Z_2}$$

where Z_1 and Z_2 refer to the impedances indicated in *Fig. 5*. This is a modification of the standard formula employed in some broadcasting plants which is $C = 100/Z_0$, where Z_0 is the impedance at any given point in a circuit. Thus, the impedance at any point on a 600-ohm line equals 300 ohms, since the source and load—each being 600 ohms—are in parallel. The value of 100 gives a roll-off which is somewhat greater than required to provide flat response.

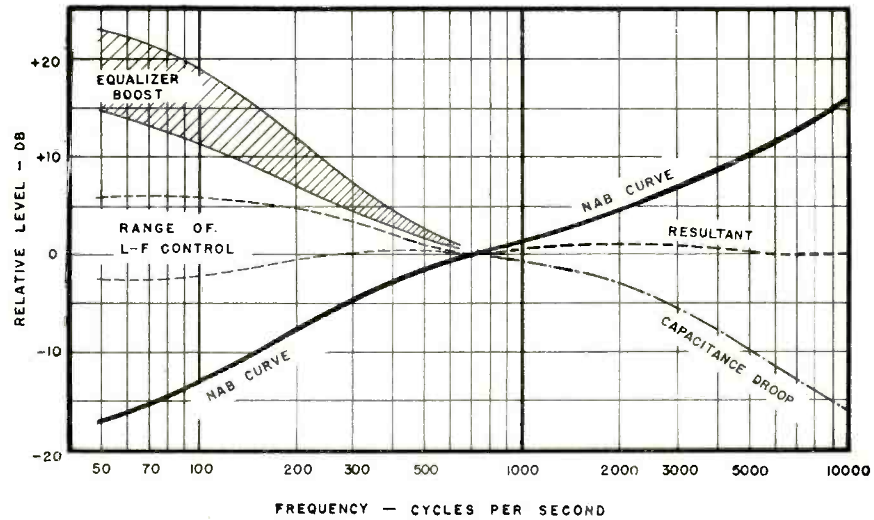


Fig. 4. NAB recording characteristic related to equalized preamplifier and to capacitance roll-off to provide control over low-frequency response and a flat high-frequency response.

Adaptations to Preamplifier

The preamplifier suggested for low-frequency compensation is readily adapted to provide the required roll-off to compensate for the NAB curve. The logical place to insert the shunt capacitor is across the output circuit, at which point the impedance consists of the load resistance shunted by the plate resistance, and both shunted by the grid leak of the following stage. Assuming that the grid leak is 1 meg. or more, the impedance at this point is numerically of the order of 40,000 ohms, resulting in a shunt capacitance value of $0.0018\mu f$.

When the connections from the preamplifier to the succeeding circuits must be fairly long, the capacitance of the shielded lead is apt to cause a droop in the response. This may be corrected easily by the addition of a capacitor across R_4 in the circuit shown in *Fig. 2* in the July issue. Assuming a value of 0.27 meg for this resistor, the capacitance to be placed across this resistor is equal to $0.59/f$, where f is the frequency at which the response is down 3 db. This same method of equalization may be used to bring the high-frequency response of the G-E pickup up to flat at any desired frequency by the application of this formula, if the particular cartridge droops appreciably. The output of Pickering cartridges is held to a tolerance of ± 2 db to 10,000 cps, and further correction should not be necessary.

The new Western Electric 1304 type reproducer is shown here being used in conjunction with Western Electric 25B speech input equipment.

Ultrasonics in Liquids

S. YOUNG WHITE

The author discusses important industrial applications of ultrasonics in liquids.

IN GOING FROM GAS to liquids in our consideration of ultrasonic energy, we find a markedly different picture. The pressures increase over fifty times, and the motion is about one-fiftieth that in gas. We cannot hope to obtain many millions of "G", but must be happy with perhaps a maximum of 100,000 or so, corresponding to about a thousand miles/per sec/per sec. for a few kilowatts per cm^2.

Provided we can generate these powers and effectively couple them to a load in a production line, what will be the desirable effects?

Emulsification

One of the most pronounced is emulsification. Practically any liquid can be so mixed with another that stable mixes are formed, especially if there is a small amount of gas in one liquid, such as mercury and milk,* aluminum and lead (in melted form), oil and water and so on. This is a high-power action, and in the clearest case of mercury and milk is rather easy to visualize. The mercury stays on the bottom, of course, by gravity separation. Intense short waves are set up on the surface of the mercury, like waves in the sea. These work up to a sharp peak in the milk, and break off in little dollops, as a sailor would say. If you try this, very often about 20% will separate after a few days, the rest remaining in suspension.

A very ingenious suggestion has been made, to use this effect for laundering clothes. The grease will emulsify into the water without soap, and the solid dirt particles will be separated by the terrific accelerations. This is the type of ingenuity we must cultivate in our thinking of ultrasonics, as the possible fields of use are limited only by our imagination.

The pronounced effects noted in emulsification teach us any two liquids of different densities will be thoroughly and quickly mixed, one probably being reduced to micron size at once, and then these particles will be violently agitated within the other liquid. This probably explains the almost universal speeding up of chemical reactions noted between two liquids. If they are of different sonic impedances, enormous work can occur at the boundary layers, with marked effect on the electric charges of the interfaces, and the production of high temperature locally in the boundary. It must be noted again that our knowledge of these boundary layer conditions is not too good in most cases, so again the best test is to try it out.

While high power will tear things up, especially with cavitation, at quite low powers the reverse sometimes takes place—colloidal size solid particles are formed into relatively large groups. This reverse effect can be looked for either at reduced power or often by going to a lower frequency at the same power as each reduces acceleration. This coagulation can sometimes be a serious nuisance, when diaphragms are used, as a coagulate forms on the diaphragm which is difficult to scrape off.

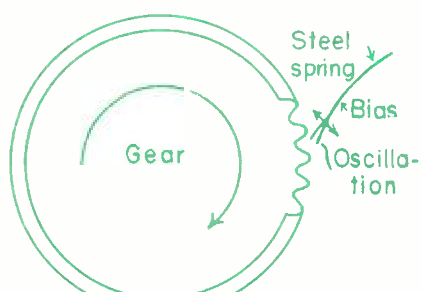


Fig. 1. Gear and biased spring.

The writer was given an interesting lesson in impedance matching while measuring the transmission loss of signal through a two-foot path of water containing about 1% paper pulp, the liquid being tap water (not degassed). Work had been done at an input signal level of about 50 milliwatts per cm^2 and checked quite well. The power was then increased to about 2 watts/cm^2. The received signal rose to a very high value for about a second, and in ten seconds fell to zero. Investigation showed that a layer of dense pulp had rapidly formed on the diaphragm. The sonic impedance of this layer was about the geometric mean between the solid magnetostriction transmitter and the water, so it formed an impedance match between them, and much more energy was coupled into the water, giving us about four times the usual signal. This energy density was sufficient to release gas from the water, however, so in ten seconds or so the gas layer became thick enough to form an almost perfect energy reflector, and the signal fell to zero. When we boiled the water, the gas layer did not form.

This coagulation effect should have wide commercial use. While it seems possible to coagulate paper pulp on a flat plate to make a very dense paper, the adhesion is so high that it would be almost impossible to remove it from the plate. If you wished to remove the particles by coagulating them out of the liquid you could continuously scrape them off a diaphragm. In some cases, the layer only forms to a certain thickness and does not increase thereafter, so you could continue to put energy into the liquid through such a diaphragm, and let the coagulates settle after they had flowed through the apparatus.

If there is considerable gas in the liquid, another interesting effect takes place. When you release pressure on such a liquid, gas will bubble out. The rarefaction portion of an ultrasonic wave will likewise release bubbles, as that portion of the wave lessens the over-all pressure locally. Now if fine particles are in the liquid, a little gas overcoat will form on each particle. If the small particle is a bacterium or yeast cell in beer, for instance, it will in many cases be inhibited and stop growing, as it must be in contact with liquid to absorb food and excrete waste. Many bacteria are unaffected by this, however, so we would have to go to cavitation to control them. Apparently no organism can withstand high-level cavitation, being torn to pieces rapidly. The gas bubble may be useful in flotation processes, however, such as ore separation. If the bubbles are allowed to escape, the liquid may be degassed, of course.

Another useful effect is that the enormous pressures and accelerations will force liquids to penetrate solids; for example, the Germans found that dyes were forced into the grains of photo-

*The homogenization of milk is an example of the emulsification power of ultrasonics, and lead and aluminum will stay mixed after cooling if "emulsified" this way. It is obvious that chocolate and many similar substances can be emulsified to give a finer grained, table mix.

graphic emulsions so they were colored to a much higher degree. Perhaps our wives can enjoy the doubtful blessing of bright purple nylon stockings. But, after all, most inventions have some drawbacks.

The Germans also teach us that grain size of photographic film emulsions can be reduced by a factor of six, which means six times the resolution in the picture. While the emulsion is being manufactured the silver haloids tend to agglomerate into groups. Ultrasonic acceleration breaks these up.

The main application of ultrasonics to metallurgy is while the melt is cooling. This is so important that we shall delay discussion until the next article which will be on solids.

Sterilization

Sterilization of food in containers is a very practical problem. Canned food is usually placed in pressure cookers and the inmost particle brought up to 250° F. or so. This breaks the continuity of the production line. This also requires girls to load and unload the rack which must be handled by an overhead crane, which then inserts it in the pressure cooker, which must then be bolted shut and steam applied. Every other operation can be a continuous process.

While some help in sterilization may come from the formation of bubbles around the bacteria, the vast majority of materials must be cavitated. Energy densities of about ten or twenty watts/cm² are required. This must be applied so that every part of the liquid is cavitated. To do this, a very confused and erratic wave pattern with multiple reflection at all angles must be used, or the high-frequency pattern must be shifted by moving the container or by frequency-modulating the generator.

In the case of a usual cylindrical "tin" can the analysis of the wave pattern is fairly simple, and we have two perfect diaphragms at either end to work with. Bottles are oddly shaped, and we may have to rotate the bottle end for end. We cannot well take more than a second to obtain our results, as we must handle several hundred a minute. It costs little to break the line into several parallel lines and then bring them back into one line again.

The type of food we can sterilize is probably limited. Simple liquids such as orange juice should be easy, but cans containing large solid objects like plums may give difficulty. We would tend to homogenize the contents so the solids would not settle.

A particular advantage is that heat is not necessary. All beer is now pasteurized at 145 degrees minimum after bottling. The taste is somewhat affected by the destruction of enzymes, so many people prefer keg beer to beer in containers. This we would avoid. There are 1800 beer lines in the country, and many thousand canning lines, so this is a good market.

We must introduce our energy over a large area of a glass container at once, as too high an energy density will cause the glass to flake off in pieces a few millionths of an inch in size. The doctors may give us glass-lined stomachs some day, but it is doubtful if they will use this method. The tin may be broken loose from the steel in the "tin" can if we hammer it too hard in one place. The can is used only once, but the bottle may go through the line ten times, so we must be more careful of the bottle. We cannot depend on the wall thickness of the bottles in calculating, as they vary two to one in thickness. Tin cans are very uniform, of course.

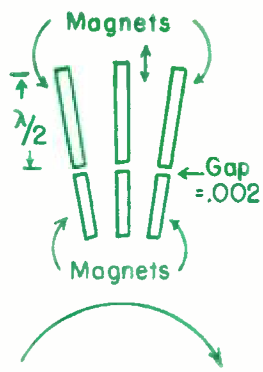


Fig. 2. Rotary magnetic generator.

We have several factors that help us in generating and handling the ultrasonic energy in liquids. From an impedance match view, we can insert a solid diaphragm in the path of the energy when we are working with liquids. The center of the diaphragm formed by the tin can will reflect only about 10 or 15 per cent of the energy, if we have water on both sides. If we hit it with sonic waves from a turbojet unit the reflection would be about 99.9%. Of course, it does not follow that reflected energy is lost—it has to go somewhere, and both steel and water have Q values in the thousands under ideal conditions. So the energy in the steel or water would simply wash back and forth as wattless component and the field strength would rise until the load took all the energy, except energy lost in the walls of the chamber.

The Q of the load is of great importance. We use Q in the radio sense—the ratio between the energy returned by elasticity and the energy lost in heat and friction. In sea water 24 kc will travel about 100,000 feet before falling to half value, so in an enclosed space with perfectly reflecting walls the resonant build-up would be very high. So, for high Q loads, the energy density in the load can be very high with rather small driving power. There is bound to be some loss in the walls of the chamber, though.

A low Q load example is cornstarch freshly wetted in water. With about 40% cornstarch the Q was about 6 at 200 kc, so the resonant build-up is small. This is the worst case in the writer's experience. It showed the energy was going into work at least in the boundary layers of the starch particles, and was probably doing something to them.

Now we have shown a few attractive possibilities of commercial use in liquids, how do we generate and apply the power?

Quartz Crystals**

Quartz crystals will give a hundred watts or so. They are a natural product and are very subject to breakage when operated at above 100 watts. You also need a one-kw transmitter to drive them, and they must be immersed in oil. As a serious power source they are useless. The PN (Primary ammonium tartrate) are more uniform, being artificial, but are rather limited in power also, because of their low dielectric constant.

Magnetostriction seems promising until you try it. At high power, the tubes crystallize and break. The magnetostriction rods must be in the form of tubes so they may be water-cooled. They must be slit for eddy currents, and the slit filled with some cement so the water will not leak out. Nickel tubes from old Edison storage batteries are pretty good. For maximum activity, heat for four hours in a hydrogen furnace at 1485° F. and let cool all night. Then if you put them in a lathe and polish them with crocus cloth they lose about half their activity, as they are very sensitive to mechanical shock.

While special magnetostrictive materials have been observed to have an efficiency of 9%, in general you must put in 30 kw to obtain 1 kw out, from many magnetostrictive tubes, and then the short life makes them a nuisance.

Let us take a look around for some other phenomena that we can utilize with the hope of securing a kilowatt or so, and with possibilities of a hundred or a thousand. It is entirely possible that the fields of application of ultrasonics are so wide that a whole series of different types may be the answer, rather than one universal type.

The simplest type to consider is shown in *Fig. 1*. This is a gear of, say, 80 teeth

**While a steel diaphragm will be cavitated by a power density of two watts /cm², quartz will operate under oil at ten watts with no sign of cavitation. This is evidently due to the much higher compressibility of oil, which must be employed in any case to provide insulation to prevent flash-over. It seems reasonable that quartz would cavitate even more readily than steel if used in water, which is of course impracticable. So oil may well be used as coupling to a bottle or other load that cannot otherwise take high energy densities.

rotating 300 rps—practically a duplicate of the rotor in the turbojet unit. We can then take a spring steel strip and hold it against the periphery of the gear. The gear teeth are only capable of giving a push, and the elasticity of the spring must provide the return part of the cycle. We do not have to try this, as common experience tells us such enormous power density will be applied to the spring the end will burn or be hammered off. The gear teeth will not be damaged very much, as the work is spread among 80 of them, but is all concentrated in a single spring.

Magnetic Units

The next obvious thing is perhaps a magnetic unit similar to that shown in *Fig. 2.* Here we place 80 Alnico magnets on a rotor revolving at 300 rps as before, and then arrange 80 similar magnets around the outside, and have a small gap, say two mils clearance. Each time the magnets line up they will pull toward each other. The idea is to allow the outer magnets to mechanically oscillate radially, make them, say ½ wavelength long so they will resonate and allow us to take the energy off their outer ends.

When we take two Alnico II bar magnets ½ inch long and one-eighth inch cross-section and measure their pull with a 2 mil gap we obtain the disappointing figure of three ounces. Since one watt per cm^2 is over 100 pounds per square inch in metal, this doesn't look too promising.

The average quarter-horse power motor runs at about two watts/cm^2 of rotor area, but some water-cooled highspeed 1600-cycle induction motors run at 40 watts/cm^2. Perhaps by going to electromagnetic fields and long rotors we could generate one or two kilowatts this way, but it is difficult to mount the outside magnets with sufficient rigidity to allow precise clearances and yet transmit lengthwise oscillations without excessive loss in the mounting. There are large lateral forces also that must be controlled, and under this type of mechanical oscillation many magnetic materials lose their magnetic properties. This looks a little discouraging at this time. The writer has the uncomfortable feeling some bright young man is going to make this work some day.

Jets

It is so easy to obtain high power densities in the turbojet unit, let us consider jets for a while. In a perfect gas orifice if we start at zero gauge pressure and gradually increase the pressure, we find that at 13.6 pounds in air we have attained sonic velocity in the jet—1100 feet/sec. No matter how much we increase the pressure, our velocity remains at that value.

Taking 15-pound pressure to allow a little for losses and imperfect orifices, we find a one inch circular jet will discharge over 400 cubic feet a minute at 15 lb. gauge. This is 21.5 h.p. adiabatic, a very respectable amount of power. Now we see what the aircraft boys are up against when they want a sonic velocity wind tunnel—50 feet diameter requires 7,700,000 h.p. at 100 per cent efficiency in the compressor, or ten million in practice.

If we consider putting liquid through the turbojet unit some rough calculation shows it would take about 300 h.p. to turn it over, with instantaneous pressure of several hundred thousand pounds per sq. in.—obviously impossible at the velocities we must use to obtain the high frequencies of ultrasonics.

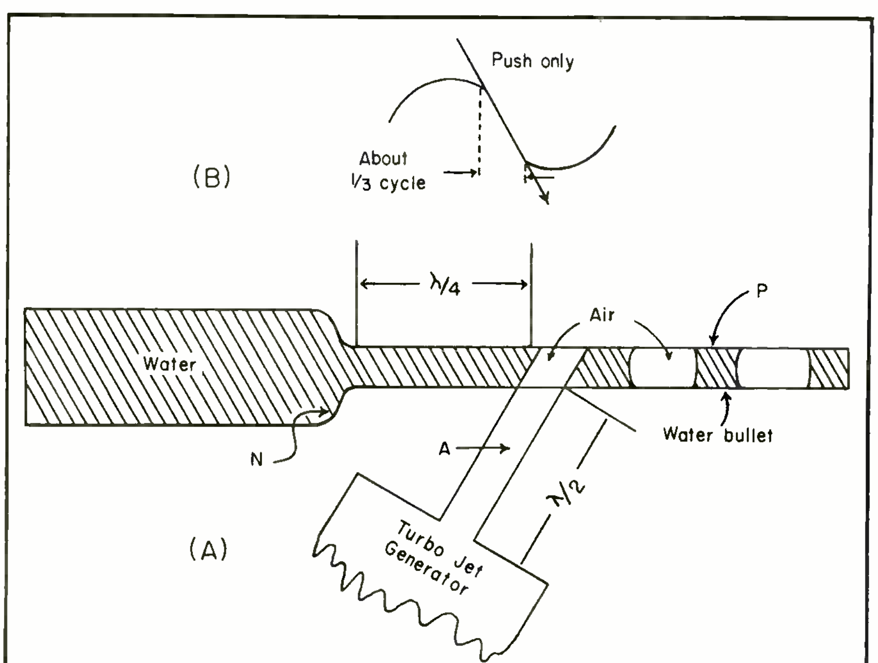


Fig. 3. A shows nozzle N designed to convert water pressure energy into kinetic energy. *B*, to inject energy into a sine wave by pushing, force must be applied for about one-third the cycle.

There is no difficulty in designing a nozzle for liquid, and little trouble pumping the liquid at a pressure of several hundred pounds to give us liquid jet velocities of several hundred feet per second from that nozzle. When we direct such a stream of liquid on a load it will of course give us a d.c., or unidirectional, push. If we could interrupt this stream at regular intervals and high frequencies, we would be shooting bullets at the load giving a regular series of pushes. The restoring force would come from the elasticity of the load, which would be the transformer necessary to turn pulsating d.c. into a.c.

We have found we cannot do this with a solid rotor, but perhaps there is another attack. We have all heard a water tap giving a 500-cycle note when at some critical point when almost turned off. We have also seen a mixture of steam and water come out of a pipe. and been impressed with the enormous energy shown in the gurgling noise it erratically gives out, and the velocity of the random slugs of water it shoots out. Perhaps we can discipline this effect in some manner.

Figure 3A shows a nozzle N designed to turn all the energy in the water pressure into kinetic energy or flow energy. This flow is continued in the small pipe P, say 60 mils (1/16th inch) diameter. The air input A is shown coming in the side, fed by the turbojet unit TJ.

The turbojet can be considered as metering or measuring a known slug of air each cycle, say 24,000 per second. This slug can be made to come out at almost any velocity and pressure we want. Thus we inject a disciplined bubble into the liquid stream each cycle.

The water and the bubble travel downstream together, the air separating the water into bullets. *Figure 3B* also shows that to inject energy into a sine wave by "pushing" only, we must push only approximately a third cycle. So the bullets must be separated by twice their own length.

This means the water comes out of the nozzle at 100 ft/sec., and after being chopped by the gas, must come out at 300 ft/sec., so about all the energy must come from the gas since the kinetic energy in the water bullet is proportional to the square of the velocity.

Concentric Injection

Since we want a rapid and definite insertion of the bubble, we can resonate both the water solid stream and the air injector pipe, so that when the wave in the water is minimum pressure, the wave and slug of air, is at maximum pressure. We should continue the water pipe far enough so the bubble can expand to twice the dimensions it had when inserted—this takes about an inch of pipe.

[*Continued on page 45*]

Design of Electronic Organs

WINSTON WELLS* PART II

In this series of articles, the author presents a thorough discussion of the design and operation of electronic organs.

THERE ARE TWO BASIC METHods by which musical tones may be produced. One is by means of oscillating systems; the other involves the use of scanning devices.

All the existing non-electronic musical instruments, as well as the vocal apparatus of animals, consist of an oscillating system with a source of energy, a filter network and some form of radiator for communicating the oscillations to the atmosphere.

In addition, most musical instruments are provided with means for varying the pitch of the sound, in order that the intervals of the musical scale may be played. This is accomplished either by varying the natural period of the oscillating system, as in the case of the flute, or by providing separate fixed oscillators for each note of the scale, as is the practice in the organ and piano.

Percussive instruments, such as the piano, generate damped waves. The hammer supplies a pulse of energy to the resonant system formed by the string under tension, and the latter continues to oscillate with diminishing amplitude until the energy is expended.

Wind instruments are continuous wave generators. String instruments, such as the violin, may usually be regarded as continuous wave generators when played with a bow, since the latter supplies a power pulse at the same phase of each oscillation of the string.

Timbre

Timbre, the complex quality which distinguishes the sound of one instrument from that of another, is the combination of effects produced by:

1. The frequency and amplitude of the partials as related to the fundamental frequency of a tone.
2. The dynamic characteristic of the tone throughout its duration.
3. The deviation from the mean pitch of the tone, throughout its duration.
4. Changes in amplitude and character of partials, throughout the duration of the tone.
5. Beats and higher order resultants, produced through the interaction of partials.

Of all these factors, the dynamics and harmonic structure of the tone are of greatest significance. (The term "harmonic structure" may be applied, even when the tone contains partials which are not harmonic.)

Aside from vibrato and celeste effects, it is seldom desirable to purposely introduce deviations of pitch in a note. Some musical instruments, particularly horns, are characteristically incapable of holding the same pitch through the inception and termination of a note, but the musician makes every attempt to do so in the majority of instances. The slide trombone and instruments of the violin family are capable of changing their pitch through a continuous spectrum, rather than by steps, but it is obviously impossible to perform these "slurs" and "slide glissandos" upon a keyboard.

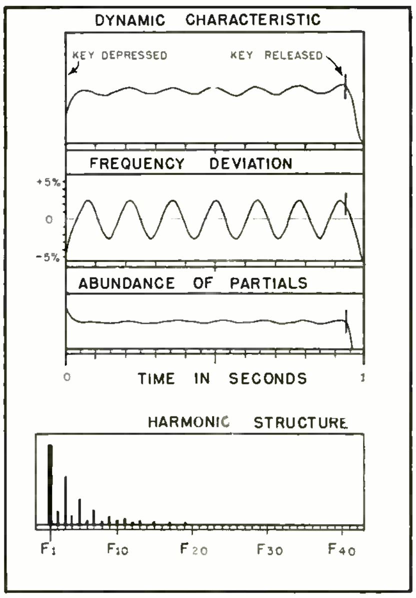


Fig. 1. Unique chart shows complete tonal character of a note, at a glance! This particular chart was made of a *tibia clausa* pipe, tuned to middle C, speaking on 15 inches of wind, and with the vibrato on. It will be noted that the frequency shift caused by the vibrato is accompanied by a concurrent change in amplitude of signal, and in harmonic content.

When attempting to imitate another instrument upon the organ, it is desirable to reproduce the effect of that instrument as faithfully as possible, yet there are limits beyond which we cannot go in practice. The violinist is free to render his artistry upon a single melodic passage (one note at a time), using both hands to shape the tone in endless ways. The organist, on the other hand, may draw a stop which will yield a tone having the *average* characteristics of the instrument to be represented, but there is little or no means for modifying these characteristics during the rendition of a passage.

It is important that we bear the foregoing facts in mind when designing an imitative stop for an organ. We must, first of all, familiarize ourselves with the character and purpose of the instrument to be imitated, then analyze the effect produced. We may then pick out the principle characteristics responsible for this effect, and devise means for simulating it. *Fig. 1* shows a convenient form of notation to use in the analysis and synthesis of musical tones.

Design Requirements

Not all of the stops on an organ are imitative of other instruments; in fact, quite a few organs have no imitative stops at all. A well-designed instrument must possess a primary group of stops which sound good when full chords are played upon them. In addition to these, it must have some stops of sufficiently interesting and contrasting tone color to be useful in the rendering of melodic passages against a background of the first group. A third group is sometimes added, the stops of which are seldom played alone but, rather, in combination with other stops for the purpose of modifying their timbre. And, finally, there are the imitative stops, which are most frequently used for orchestral effects.

Since the organ is most often played with several stops speaking simultaneously and, since the full organ is necessary for climatic effects, it is of utmost importance that all except the solo stops be capable of blending in tone when played in unison. This subject will be discussed in detail at a later date. For the present, it will suffice to say that a good ensemble is most easily attained through the blending of tones whose partials are true harmonics, the partials diminishing in amplitude with their order.

It is apparent from the foregoing discussion that there is no element in the production of musical sounds by accoustical means which cannot be simulated electrically. Furthermore, the electrical approach gives us considerably greater control over the characteristics of the tone producing elements, since we may

*307 E. 44th St., New York City

work with relatively pure circuit elements (inductance, capacity and resistance) rather than lumped circuits. Whether we build an exact electrical counterpart of an acoustical instrument, or whether we utilize a novel circuit to yield the same tonal effect, is more often a matter of convenience and economy than of physical necessity.

In the case of non-electronic musical instruments, the basic signal is generated by an acoustical or mechanical oscillator, its harmonic structure being modified by the resonator, which behaves as a filter network and, usually, as an impedance-matching device between the oscillator and the atmosphere. The violin is an ideal example of such a system. The flute is an instrument in which the resonator serves as the oscillator tank circuit, filter network and radiator, all in one.

We may produce a perfect sustained violin tone by feeding the output of a vacuum tube oscillator into an electrical filter, then into an amplifier and loud speaker. We may provide an ideal vibrato by modulating the oscillator plate supply at about seven cycles per second. We may key the note by means of a switch, placed anywhere in the audio circuit although, with most oscillators, greater realism may be attained through keying the oscillator plate supply. This takes the "edge" off of the attack and decay of the tone, and more closely simulates the bowing of a violin.

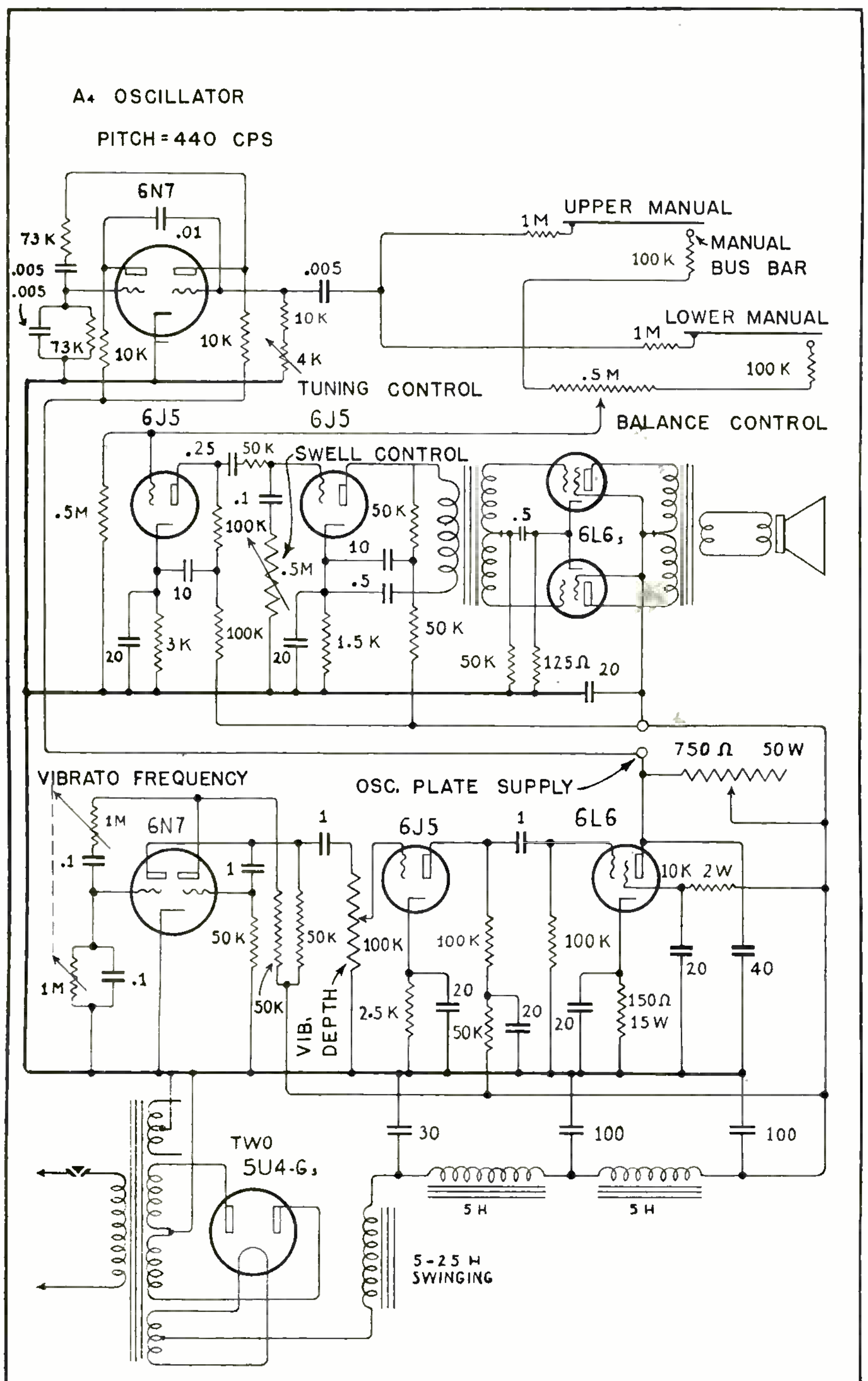


Fig. 2. Circuit diagram of an experimental electronic organ.

If a number of these oscillators, (tuned to the steps of the equally tempered scale), are coupled to the filter and amplifier system, and if they are keyed through contacts linked to the keys of an organ manual, we have an excellent violin stop. Melodic and harmonic passages, played upon such an instrument would be indistinguishable from identical passages rendered by violins, as long as we restricted ourselves to "straight fiddle playing" and avoided techniques incompatible to the keyboard.

Figure 2 shows the circuit for a modified instrument of this type. The values of the components were taken directly from one of our experimental models and, while operative, do not necessarily represent the ideal.

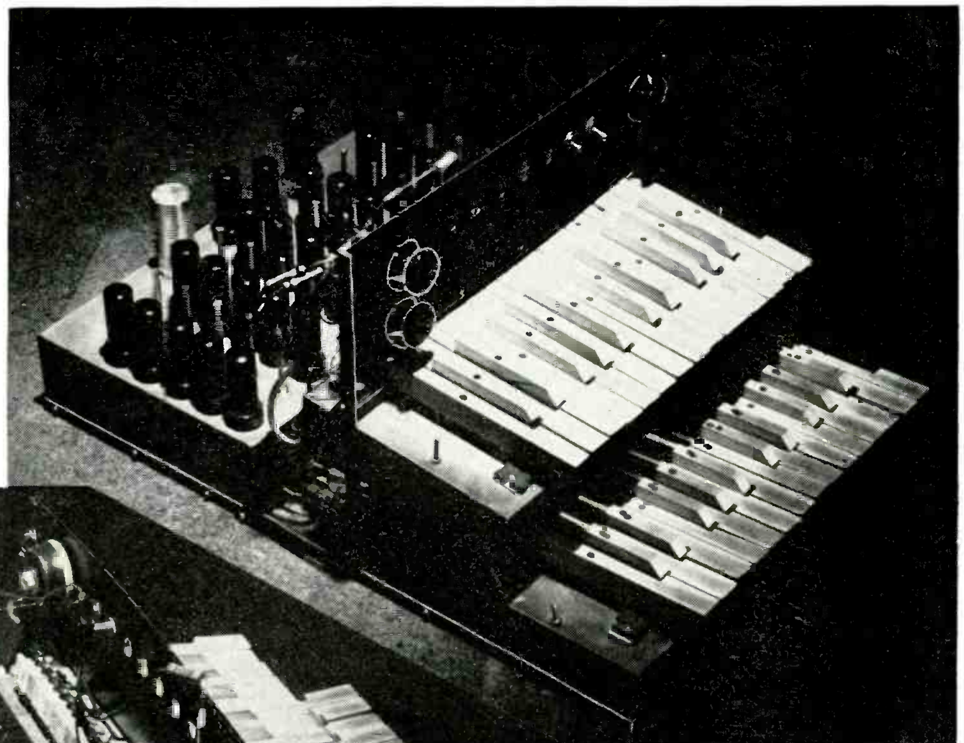
The oscillators are resistance-capacity tuned, and the output is keyed instead of the plate supply, since it was desirable, in this case, to form a separate stop for two manuals from the one rank of oscillators. Only one oscillator is shown, the others being coupled through their respective key contacts to the manual bus bars, which are common to all the keys of one manual. The filter, corresponding to the body of the fiddle, is a simple resistance-capacity network of the "tone control" variety, and it serves as the coupling between the first and second stages of the audio amplifier.

The Vibrato

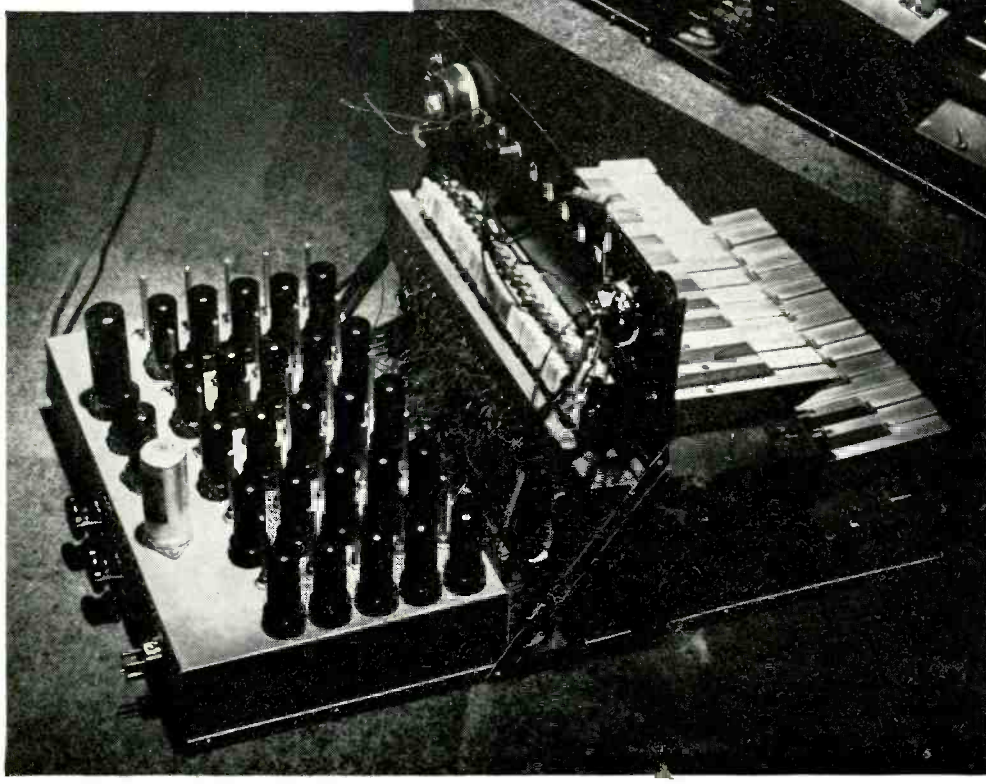
The vibrato consists of a low-frequency oscillator, followed by a buffer amplifier which drives the power stage. The plate of the power output tube is connected to the common plate supply terminal for the tone generating oscillators, the vibrato output forming a variable shunt across the power supply of the tone generator. Sufficient resistance is introduced, between the main power supply and the terminal feeding the tone generators, to enable the swing of the vibrato output to modulate the power fed to the tone generating oscillators.

The frequency of the vibrato oscillator is continuously variable from four to twelve cycles per second. The amplitude of the beat is continuously variable to give a maximum frequency deviation of about seven per cent to each side of the mean tone generator frequency. The most pleasing effects for general use are obtained with the vibrato set to seven cycles, with a frequency deviation of two to three per cent.

In the use of vibrato, it is important that the frequency of the note be shifted by the same percentage of deviation to each side of the nominal frequency, since the ear fixes the pitch of the note at a point, midway between the extremes. Any departure from this practice will result in an apparent change in pitch, when the vibrato is turned on or off.



Photograph of two-octave, two-manual experimental organ (circuit shown in Fig. 2). The assembly includes entire instrument with the exception of the power supply and amplifier. Knobs on front left or panel control balance of volume between manuals and high frequency cutoff, respectively. Knob at extreme right regulates oscillator plate potential, permitting the pitch of the instrument as a whole to be raised or lowered.



Rear view of the experimental model, showing chassis layout. The shafts between the rows of tubes are for the tubing controls, one for each oscillator. The two knobs on the rear of the chassis control the vibrato frequency and and the depth of beat, respectively.

It should also be noted that vibrato circuits of the type described can only be used with tone generators whose frequency is a function of the applied voltage.

Circuit Design

Figure 3 shows the essential circuits of a complete electronic organ, designed by the author. The instrument, as shown, is provided with five stops, available on both manuals and the pedals, in addition to octave and sub octave couplers.

All the stops are derived from a single rank of oscillators, a separate filter being used on each note, for each stop. Through the proper choice of filter circuits and constants, it is possible to reject or accept the partials in the oscillator's signal in any combination desired. This gives us the choice of a wide variety of tone colors, since the signal from the oscillator used is exceptionally rich in upper partials.

There is no theoretical limit to the number of stops which may be derived from a single rank of oscillators in this fashion. However, changes in loading affect the pitch of these oscillators, and it becomes increasingly difficult to maintain constant loading as more stops are added. This difficulty might be surmounted through the use of buffer amplifiers or constant impedance keying devices, but the additional cost would be better justified, if used to provide an extra rank of oscillators.

This would permit the use of separate vibratos for the two sets of stops, and would yield a better ensemble on full organ. Furthermore, the two ranks of oscillators could be adjusted differently, to provide partials of distinctly different types, thus providing further contrasts of tone color. It would be highly desirable to use five or more ranks of oscillators in instruments where expense and bulk could be disregarded.

The borrowing of stops from one manual to the other is a bit reminiscent of the practice of duplexing stops on the pipe organ. There is nothing objectionable about it, however, when used in the manner indicated, since the stop may be had at different strengths on different manuals, and the corresponding notes on separate manuals are additive in strength, when played together.

The octave and sub-octave couplers also differ from those found on the pipe organ, in that their notes are additive, when duplicated upon the manual. The couplers are derived through the use of three contacts under each key. One is connected to its nominal note, another to the octave above and the third, to the octave below. It would, of course, be possible to couple any other intervals together in a similar fashion, as well as various registers of individual stops, thus unifying the instrument.

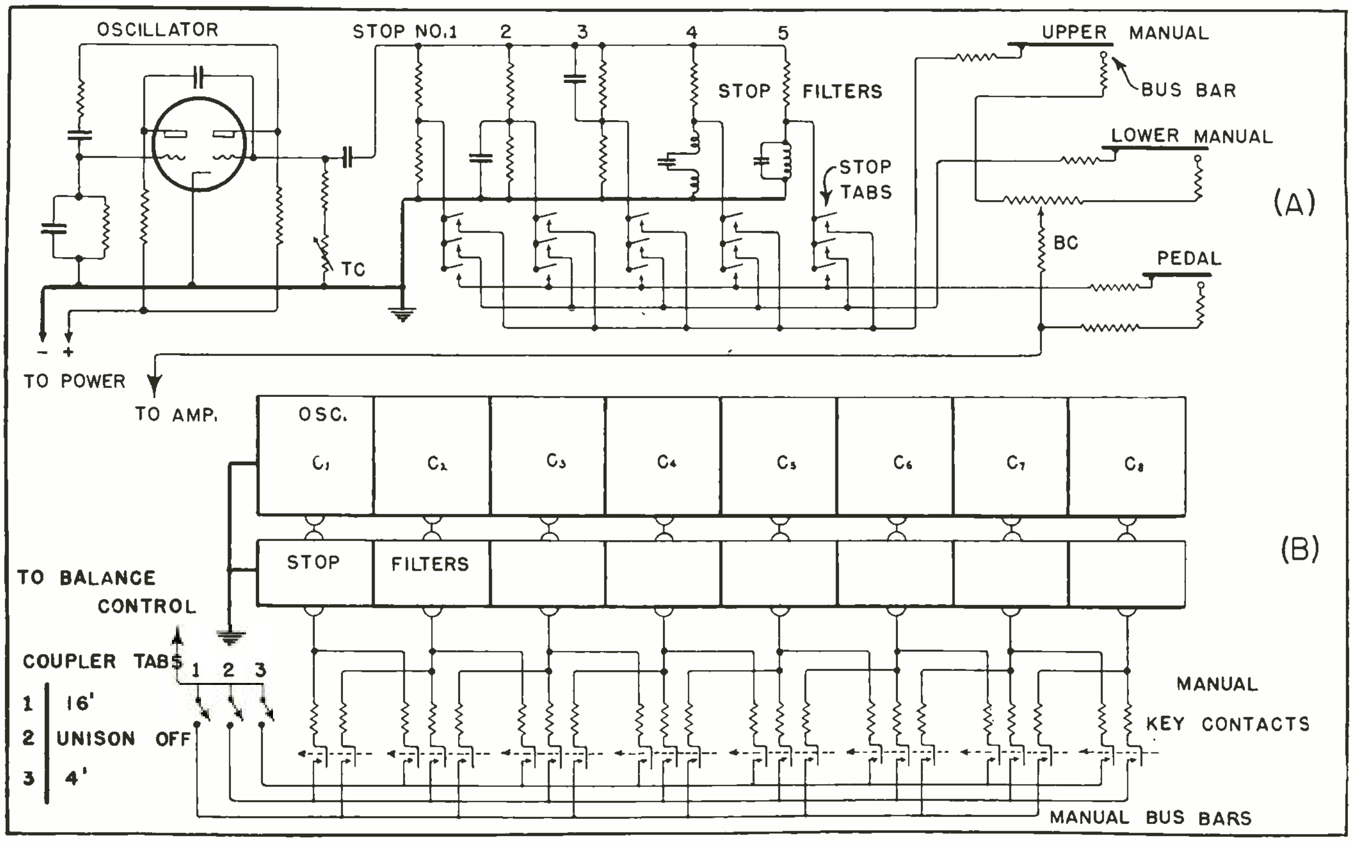
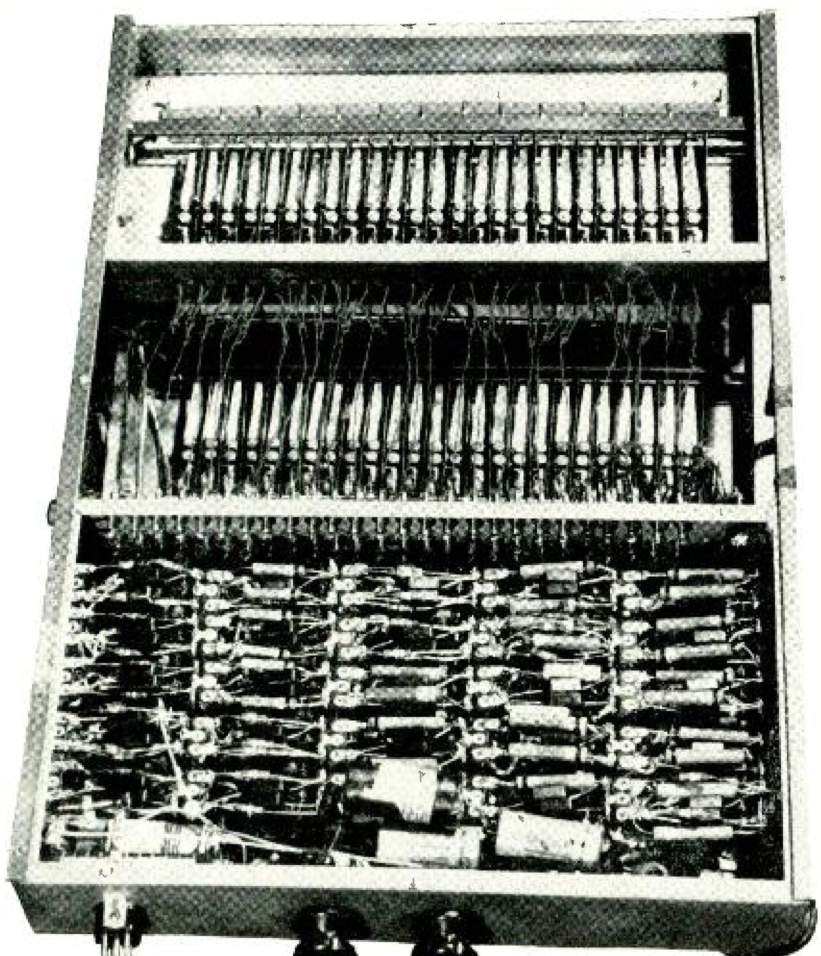


Fig. 3. Essential circuits of complete electronic organ, designed by the author.



Bottom rear view of chassis, showing layout of components. Resistors and condensers are mounted in tiers on movable frames, permitting access to those underneath. The manual key contacts are shown. It is necessary to use electrostatic shielding between all contacts, since the associated circuits are of extremely high impedance.

Frequency Drift

Oscillators of the type used in these experiments are subject to considerable frequency drift, unless stringent measures are taken against it. The power supply must be well regulated, and the circuit components mounted rigidly. The resistors and condensers must either possess a near zero temperature coefficient, or be graded in sets, the condensers having an equal and opposite temperature coefficient to the resistors, over the normal range of operating temperatures. The oscillator tubes must also be tested for stability, since changes in transconductance affect the frequency.

The performance of organs of the type discussed compares favorably with that of the better pipe organs, as far as tonal resources and quality are concerned. Instruments with but a single rank of oscillators are lacking in ensemble, but so are the smaller type organs. When several ranks of oscillators are used, a grande celeste is produced which equals that of the largest orchestra or pipe organ.

The continuously variable vibrato offers a considerable improvement over anything encountered in the pipe organ field. Numerous attempts were made to develop a device of this kind, but none that we have heard of were entirely successful.

It would be absolutely impossible to duplex the stops of a pipe organ in such a manner that the corresponding notes on different manuals would be additive.

[*Continued on page 44*]

A-C Voltage Stabilizers

LEO L. HELTERLINE, JR.*

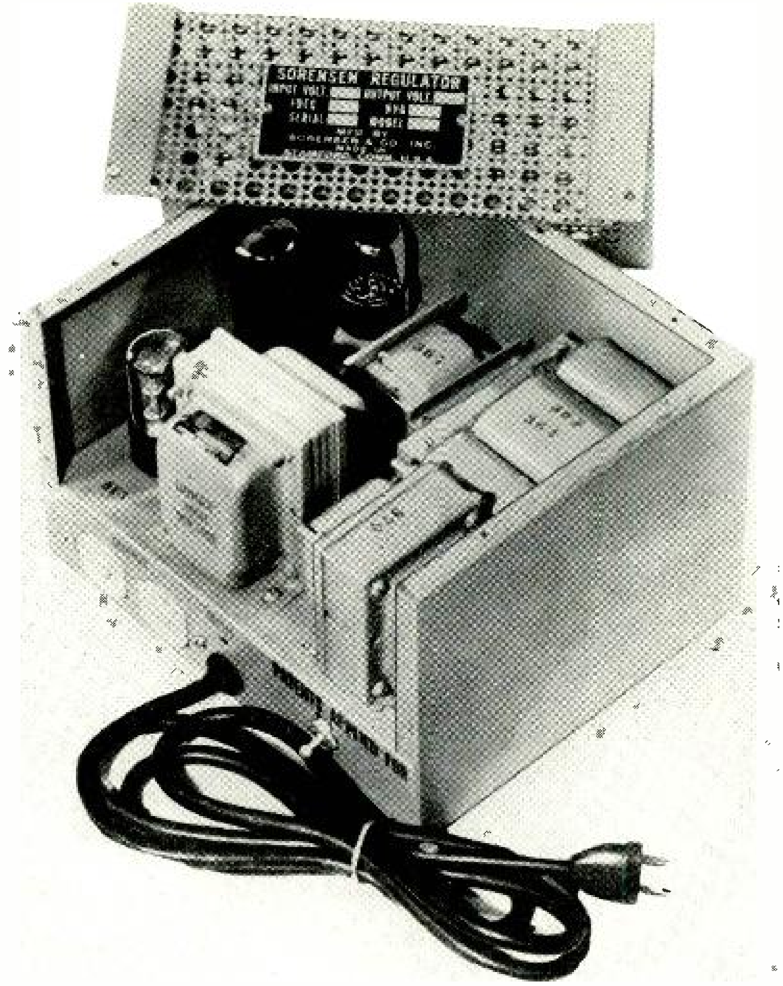


Fig. 1. The Model 150 line-voltage regulator, designed for mounting on a standard rack panel or to be used in its own housing. This unit provides good regulation at low cost for a wide range of audio and allied equipment.

IT NEVER DOES an engineer any harm to examine his assumptions. One of these assumptions which is often made is that audio equipment will be used under the conditions prevailing in the laboratory, or under conditions prevailing in the particular manufacturing or processing plant in which this equipment is made.

A good example of the effect of changed conditions is presented by a study of distortions. Normally operated high-quality amplifiers are rated at 1 or 2 percent distortion at a line-voltage input of 117 volts. Reducing the line voltage by 5 to 10 volts will often more than double these distortion figures.

Another assumption is made regarding pure waveforms. Most a-c lines have a reasonably good waveform, unless highly reactive loads or those with transient peak requirements are supplied from the same transformers. A good waveform is a necessity when certain types of voltage-multiplying circuits are used, since the performance depends upon the crest value of the line voltage. Such circuits are limited when the waveforms are markedly altered, and the effects of surges are more deleterious than the gradual fluctuations that are experienced in a normal line. Generally speaking, most audio amplifiers do not require precision regulation of voltage. However, to attain the ultimate in performance, voltages must be held within four or five volts of their design center.

D-C Amplifiers

D-c amplifiers used in laboratory work and in many other modern applications of electronics are perhaps the most critical of supply voltages, and many such amplifiers are completely unworkable unless their filament supplies are fed from sources comparable to batteries. Where motor speeds are related to voltage, the control of that voltage again becomes important. Equipment destined for the foreign market must be planned to be as free as possible from the effects of changing voltage, as well as frequency, at the input terminals. It is not sufficient that some form of voltage regulation be supplied which is workable at the nominal frequency but which becomes completely unregulated if the frequency shifts appreciably.

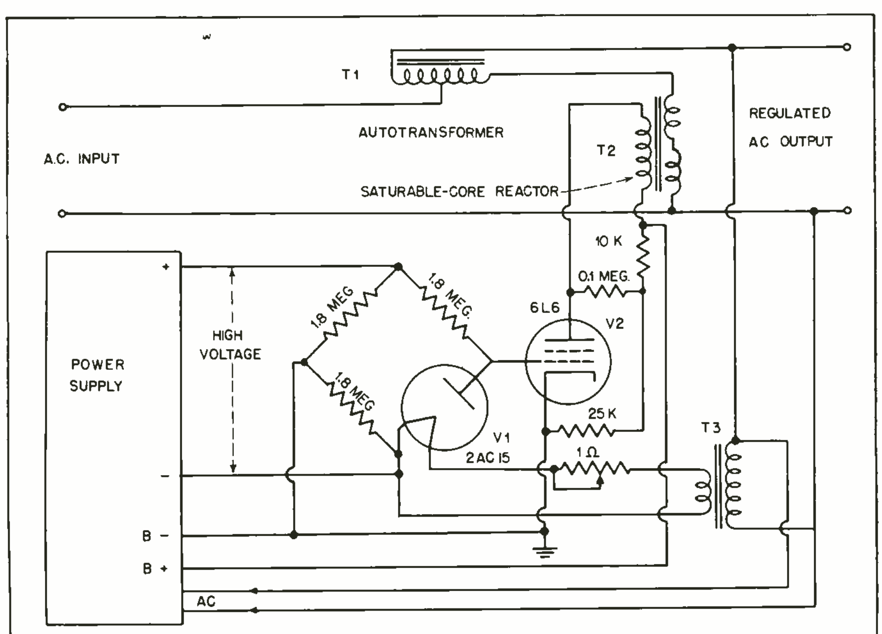


Fig. 2. Schematic of Model 150 Sorenson regulator. V_1 is special diode used as the voltage-sensitive control for the 6L6 current through a saturable reactor.

*Chief Engineer, Sorenson & Co., Inc., Stamford, Conn.

The solution to the operation of electronic equipment on lines furnishing variable voltage *and* variable frequency is to use some form of voltage regulation which is independent of frequency. Needless to say, the regulator which will fulfill these requirements is also capable of functioning correctly on lines supplying constant frequency, so that it is not necessary to make any change in equipment even though the frequency of the supply be changed, provided the design of transformers is adequate for the lowest frequency encountered.

One such regulator is manufactured by Sorenson & Co. Inc., and the degree of control is well within the limit specified by very critical applications. The regulation curve of a typical line-voltage regulator is shown in *Fig. 4*, and the schematic of the regulator is shown in *Fig. 2*.

The voltage-sensitive element of this device is a special filamentary diode, also manufactured by Sorenson for this particular use, and held to close limits. This tube resembles an ordinary vacuum tube, but has some unique characteristics. The filament is of pure tungsten, and the vacuum is considerably higher than is usual with ordinary radio tubes. The principal characteristic of importance is that the plate resistance is a highly retraceable function of the filament voltage. *Fig. 3* illustrates the appearance of the tube itself.

Diode Operations

To understand the action of this diode, assume a change in the output voltage in the upward direction, resulting from a reduction in load or an increase in line voltage. This change causes an increase in the filament voltage, and moves the operating point along the plate resistance-filament voltage curve in such a direction that the plate resistance is lowered by an appreciable factor. This plate resistance forms a portion of a bridge circuit, and the unbalance voltage is fed into the grid of a beam power tube. The beam tube draws its plate current through the d-c winding of a saturable reactor, the a-c coils of which act as a variable reactance in the line circuit.

From an inspection of the a-c circuit, it is seen that the voltage appearing across the output terminals is dependent upon the line voltage, the step-up ratio of the autotransformer, and the reactance of the a-c-coils of the saturable reactor. As V_2 draws more current, the reactance of the a-c coils is lowered, and consequently

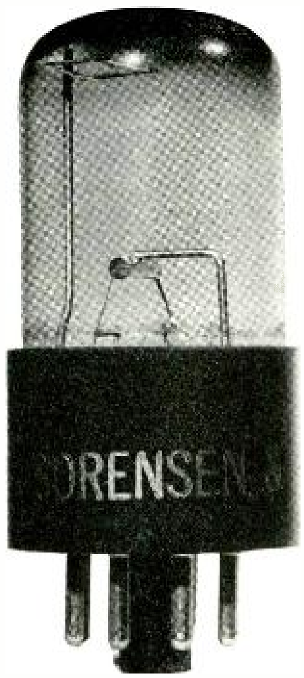


Fig. 3. A close-up of the filamentary diode, Sorenson 2AC15. The control function depends upon the plate resistance of this tube, which may be energized by either a-c or d-c voltages.

a higher voltage is applied to the autotransformer. The converse is also true, and the bridge circuit may be so connected that the change in the plate resistance of V_1 serves to compensate for line voltage changes.

The gain of this particular arrangement is of the order of 50,000 when measured as the ratio of voltage change across the d-c portion of the saturable-core reactor divided by the change in output voltage. This degree of control is sufficient to maintain the output within much less than 0.5 per cent over a wide fluctuation of supply line voltage and load current.

Proper design can also compensate for the resistance of the leads from the regulator to the load. This is particularly important when the load fluctuations are great and the leads are long. Thus, it is possible to so arrange the controls that the voltage at the *load* remains constant, even though the output voltage changes slightly in a reverse direction when measured at the output terminals of the regulator.

Since there are no resonant elements in this regulator, the action is completely independent of frequency over a range of plus or minus 30 per cent or better.

The manufacture of the tungsten filament diodes is a critical operation if the best possible characteristic is desired. Since the diode is the true voltage reference, it must be independent of all factors except the voltage across the filament of the tube. Other possible sources of dependency arise from thermal ambients, change in filament emission with time, and changes in other characteristics with time. All these problems have been investigated in the manufacture of this type of diode.

It has been found that high vacuum is necessary to produce a truly stable source. Compensation for ambient temperature is accomplished by careful design of the elements and the correct choice of metals with respect to their expansion coefficients. The order of stability of these units constitutes an improvement over voltage regulator tubes of the gaseous discharge type by a factor of at least 10. The life characteristics of these units when employed at low plate-current values exceeds that of practically any commercial vacuum tube. A simple analogy of life comparison is that of running an incandescent lamp at approximately one-half its rated voltage. As established by Langmuir and others, life is a multiple-powered function of filament temperature.

Stability is of utmost importance and can only be obtained by careful techniques. Assume, for example, that there is a plate voltage of 500 volts across the diode elements. A change of 0.1 volts in the filament supply to the diode will produce a change of 300,000 ohms in the plate resistance of the tube, and if this tube is in series with a 1.0-megohm resistor, it will change the plate potential approximately 100 volts. If this change is impressed on the grid of the beam power tube, it is obvious that it will more than control the tube over its full range of grid voltage characteristic.

Rectified A-C Supplies

This type of circuit is also adaptable to rectified a-c supplies employed for furnishing filament and relay voltages. Since the sampling voltage may be either

[*Continued on page 43*]

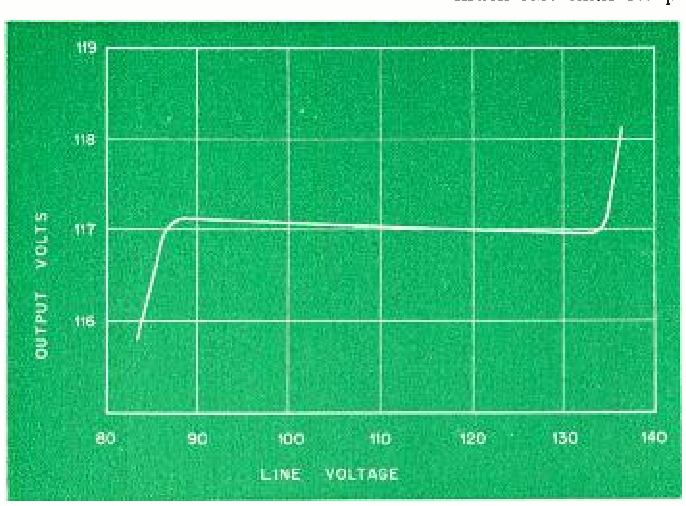


Fig. 4. Input vs output voltages of Model 150 voltage regulator. Slope of regulated portion of this characteristic may be varied to compensate for resistance of leads from regulator to load.

Planning a Studio Installation

J. D. COLVIN*

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

BEFORE AN INSTALLATION can be made and put into operation a considerable amount of material must be purchased. The greater the amount of this material that can be purchased and had on hand before the construction actually begins, the faster the work will proceed and the least amount of temporizing will take place because the exact item required cannot be obtained at the moment. A complete list of materials required for the rack and console can be obtained, since these have already been laid out in detail. The total amount of conduit and a good approximation of cable for the installation can be determined from the conduit layout. There are, however, a number of items that do not show on the diagram and layouts made so far that must not be overlooked and a list of such items follows as they occur to the author.

Additional Items Needed

Microphones—velocity, pressure, directional; microphone stands—floor, desk, boom, hand; microphone plugs, receptacles and extension cords; turntables, tubes for all amplifiers, spare tubes, spare fuses; clocks, touch-up paint to match the equipment finish; cable shield termination (such as Thomas and Betts), loudspeakers—talkback, monitoring, office speakers, remote gear-amplifiers, mikes, stands, cables, carrying cases; headphones, patch cords, attenuater cleaning solutions, work bench and tools, circuit check meter, telephone equipment—lines, instruments. There may be and probably are other items that must be taken care of before the job is 100% complete, but if all of the above are considered, the remainder will be very minor (or so obvious as to be over-looked until required).

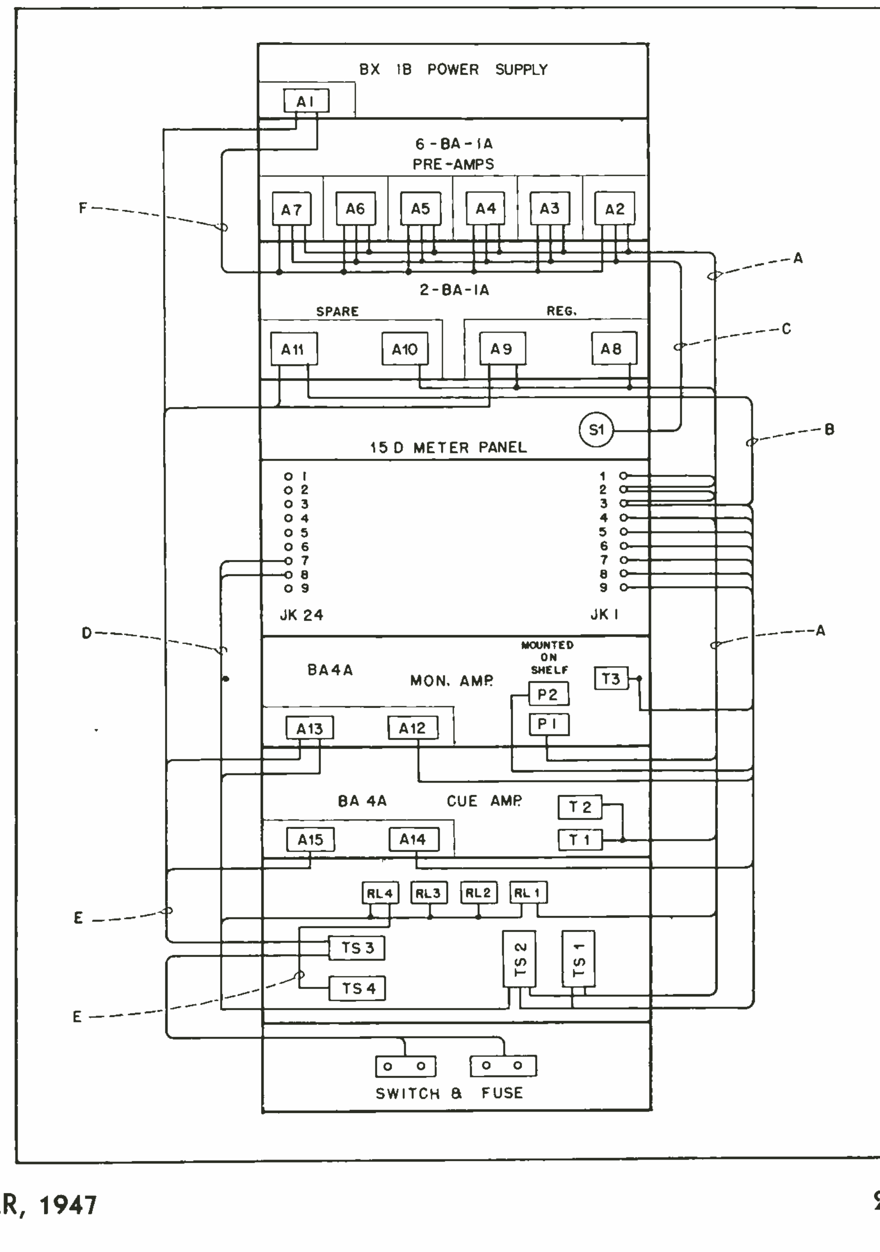
Having completed the mechanical layout of the equipment in the rack the next step is to prepare information in some form that will guide a wireman to connect the individual components to the proper points, as determined by the block diagram. Not only must the exact electrical connections be shown, but cable runs containing circuits within the same level groups as used in the conduit layout must also be shown. Nothing should be left for the wireman to figure out in the way of circuits or levels—his only contribution to the finished racks should be perfect soldered joints and neatly formed, tightly laced cables.

Running Sheets

A very convenient form in which the wiring information can be compiled is known as "running sheets." These sheets list every terminal to which a connection is made and indicate the connections to be made from terminal to terminal. By making the connections indicated on the running sheets line by line, the rack becomes completely wired when the last line is checked off.

Before the running sheets can be compiled, it is necessary to prepare a pictorial sketch of the rear view of the rack showing the location of all terminal

Fig. 1. Pictorial sketch of the rear view of the rack.



*Audio Facilities Engineer, American Broadcasting Co.

	FROM			TO		
Pair	Equipment	Terminals	Cable	Equipment	Terminals	Circuit
1	Jack row 1	1T	A	TS 2	10-1	LL Trunk 1
	"	2T		"	10-2	
2	"	3T	A	TS 1	1-1	Mic 1 out
	"	4T		"	1-2	
3	"	5T	A	"	1-3	Mic 2 out
	"	6T		"	1-4	
4	"	7T	A	"	2-2	TT 1 out
	"	8T		"	2-2	
5	"	9T	A	"	2-3	TT 2 out
	"	10T		"	2-4	
6	"	11T	A	"	3-1	TT 3 out
	"	12T		"	3-2	
7	"	13T	A	"	3-3	Mixers out
	"	14T		"	3-4	
8	"	15T	A	P1	A out	Divide pad reg out
	"	16T		"	"	
	"	17T				
	"	18T				
9	"	19T	A	P1	B out	Divide pad emg out
	"	20T		"	"	
10	"	21T				
	"	22T				
11	"	23T	A	RL 1	3	Mic 1 relay out
	"	24T		"	6	

Fig. 2. Form used in making running sheets.

blocks on amplifiers, transformers, pads, etc., and a designation for each terminal block assigned. Such a sketch is shown in *Fig. 1*. It will be noted that the terminal blocks on the amplifiers are designated as *A*1, *A*2, etc. Transformers are referred to as *T*1, *T*2. Terminal strips carry the designation of *TS*. Pads are *P*. Jack rows are numbered from top to bottom and the jacks in each row are numbered from right to left as viewed from the rear. (This gives a normal left-to-right numbering from the front of the rack which is of greater convenience for operation.

There are sometimes components that do not have a definite, stenciled terminal board, such as rotary selector switches, lever keys, etc. In such cases it is well to make a separate sketch of the component in sufficient detail and assign terminal numbers to the various solder points. These sketches can then be checked by the wireman while working on the rack to determine the particular point on the unit being referred to in the running sheets. The most commonly used method of numbering terminals on standard 40 or 60-pair telephone type of terminal blocks that are used in making the external connections to the racks is to refer to each row by number counting from top to bottom (usually 1 to 20) and the terminals in each row by number, counting from front to back (usually 1 to 4 or 1 to 6). With this arrangement, any particular terminal is referred to by row number and its number in the row. For example, the very first terminal on the block would be 1-1. The very last terminal on a 40-pair block would be 20-4. Jacks are referred to by row number, the number of the jack in the row and whether the connection is to be made to the tip or normal of the jack. A standard jack row consists of twelve pairs of jacks but each individual jack in the row must be numbered from 1 to 24 in order that the proper phasing of circuits can be maintained throughout the entire wiring of the rack. For example, the tips on the first pair of jacks in the first row would be *Jack Row 1, 1T* and *2T*. (*T* standing for tip). The last normals of the last pair in the second row would be *Jack Row 2, 23 N* and *24 N*. Although *Fig. 1* shows all cable runs in place, these are not drawn in at the time the terminal blocks, etc., are located but are developed as the running sheets are prepared, and it is determined how many of these are required and where they go.

External Connections

Of all the terminal blocks shown on the rack sketch, the only ones at which a choice of terminals can be made are those used to make external connections to the rack. In this case they are *TS1, 2, 3* and *4*. All others are definitely determined by the apparatus of which they are a part and are numbered by the manufacturer of the equipment. Since there is a choice, a definite attempt should be made to group terminals for circuits within the same level grouping as used previously. No trouble from cross-talk should be experienced by running cables of low, medium and zero levels to the same terminal block. High levels and control circuits should be placed on a separate block. Variations to this arrangement are permissible if several rows of unused terminal are left between level groups of more than 30 db. It is also advisable to attempt as far as possible to keep the normally used program circuits on the same terminal block. These would be such circuits as microphone outputs, preamp input and outputs, faders in, mixers out, program amplifiers in and out, line in, etc. Other miscellaneous or seldom used circuits as trunks, spare jack circuits, control circuits and spare terminals for any future additions should be put on a separate block. The purpose of such segregation is to keep the block having the permanent connections mentioned free from those circuits most likely to be changed and where additions might be made. The less a terminal block is worked on after it is once put in the less likelihood of trouble from broken or shorted wires later on. This is not a hard-and-fast rule since often the number of terminals used up by the program circuits are so few it would be foolish to put in a second block to carry what odd circuits that might exist. A general rule to follow is to start at the top of the terminal block with the low-level microphone and turntable circuits and end up at the bottom of the block with the higher levels. Also, where groups of similar equipment are used, such as microphones, preamplifiers, etc., connections to the terminal block should run in consecutive order as *Mic 1, Mic 2, Mic 3* or *Preamp 1 In, Preamp 2 In*, etc.

Since the jack field is the most common point of all circuits in the rack it is the most logical place to start the running sheets. The form used in making the running sheets is shown in *Fig. 2*, and since it is sheet 1 it indicates all the connections from the jack tips of *Jack Row 1*. Connections are being indicated as going from the jacks to their respective designations. Thus the first step is to list under the "From" equipment and terminal columns all terminals in the rack starting with the jack rows and the remaining equipment listed alphabetically—as *A* for amplifier, *P* for pads, etc. The next equipment to be listed after the last jack row would be the amplifier whose terminal block is designated on the rack sketch as *A1*. All terminals on the *A1* block would be listed in numerical order. *A2* would follow with its terminals and so on. A total of 24 pages were required to list all terminals for the rack under discussion.

Terminals

Having listed all the terminals from which connections are to be run, the next step is to determine and list the equipment terminal blocks and terminals on the blocks under the "To" column. This work will determine the actual wiring and placement of the wiring in the rack and it is necessary to have a copy of the block diagram and the jack field lay-out at hand as the source of circuit information. An examination of the jack field shows that the first pair of jacks or terminals *1T* and *2T* in the first jack row are for a low-level trunk. As such it will be connected to a terminal strip at the bottom of the rack. However, as it was discussed previously, it is desirable to start at the top of a terminal strip with microphone connections and therefore the terminal designation for this and similar pairs of jacks will be deferred until the main program carrying circuits in the jack field are finished.

The next six jack pairs are microphone, turntable and mixers outputs and go to the terminal block. They are therefore assigned to *TS1, 1-1* and *1-2* up to and including *3-3* and *3-4*. The title of each circuit as *Mic 1 out, Mic 2 out,* etc. is filled in the column headed "Circuit," and are for future convenience in locating terminals. The next jack pair having terminals *15T* and *16T* are indicated on the jack field as being *"Divide Pad Reg. Out."* This jack pair on the block diagram shows it being connected to the output of the divide pad feeding the input of the regular program amplifier and the rack sketch shows this pad as being designated *P1*. Accordingly, *Jack Row 1, 15T* and *16T* will be indicated as being connected to equipment *"P1"* and the *"A out"* terminals. Similarly the remaining connections are determined and are recorded. It will be noted that terminls *17T* and *18T* as well as *21T* and *22T* of *Jack row 1* show no connection. This is due to the fact that these jacks are multiples of the divide pad regular and emergency outs. Such multiple connections on the jack field can be shown on the same type of form as used in *Fig. 2*, or by a sketch shown in *Fig. 3*. Use of the sketch seems to be preferred by wiremen since it will show all normal and multiple connections pictorially on one sheet whereas possibly six or more of the form sheets would be required.

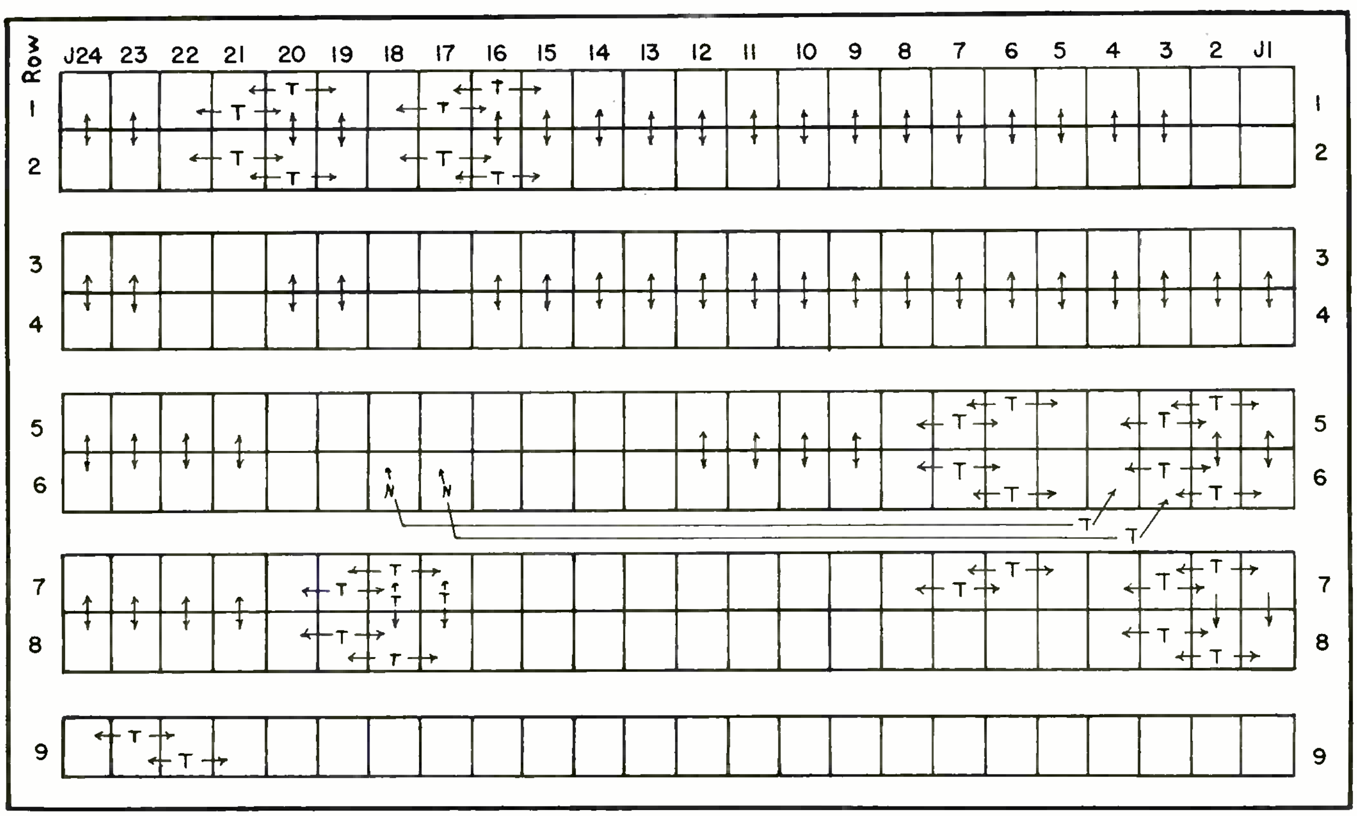
Numbering

Referring again to *Fig. 2*, the pairs should be numbered consecutively in the "Pair" column. This pair numbering will be found useful in finishing the running sheets and is sometimes used by the wireman in marking pairs in cables. Indication of the "Cable" in which each pair is to be run must now be indicated, both on the running sheets and on the rack sketch. It will be observed from a study of the block diagram that all pairs listed on this first of the running sheets come within the limits of the "Low" level grouping and therefore can go in the same cable. Since it is convenient to label cables alphabetically, the letter "A" is therefore assigned and entered in the column head "Cable." On the rack sketch, a line starting at jack row one is run down to *TS1*. This line at the moment represents the cable consisting of the two microphones, the three turntables and the mixer out circuit. However, the running sheet shows two pairs going to *P1*, and therefore a turnoff from the line already drawn will be run over to *P1*. Likewise, a line will connect cable *A* to *RL1*. The additional connections into the *A* cable as shown in *Fig. 1* over those mentioned above are the result of other pairs being assigned to this cable in latter running sheets. In this manner, the running sheets are built up and the cable runs added to the rack sketch.

Other Checks

When each pair of terminals is indicated as being run from one equipment to another, it is advisable to go to the sheet on which the "To" equipment is listed and indicate that those terminals are to be run back to the point from which they came. For example, on *Jack Row 1, 1T* and *2T* are shown going to *TS2 10-1* and *10-2*. In *Fig. 4*, the sheet showing that portion of *TS2* which contains *10-1* and *10-2*, it will be noted that it is shown going to *Jack Row 1, 1T* and *2T*, that it carries the same pair number, cable letter and circuit name. It will also be noted that an asterisk is placed in the box containing the cable letter. This asterisk is to indicate to the wireman that this pair has already been run in and should not be duplicated. An examination of *Fig. 4*

Fig. 3. Sketch showing multiple connections required in the jack field.



	FROM			TO		
Pair	Equipment	Terminals	Cable	Equipment	Terminals	Circuit
73	TS 2	8-1	B*	Jack row 8	11 T	Spare
	"	8-2		"	12 T	
74	"	8-3	B*	"	13 T	Spare
	"	8-4		"	14 T	
75	"	9-1	B*	"	15 T	Spare
	"	9-2		"	16 T	
	"	9-3				
	"	9-4				
1	"	10-1	A*	Jackrow 1	1 T	LL Trunk 1
	"	10-2		"	2 T	
12	"	10-3	A*	"	3 T	LL Trunk 2
	"	10-4		"	4 T	
70	"	11-1	B*	Jack row 8	1 T	PGM line to MC
	"	11-2		"	2 T	
71	"	11-3	B*	"	5 T	Spare
	"	11-4		"	6	
	"	12-1		RL 3	7	Intercom relay control
	"	12-2		"	9	
	"	12-3				
	"	12-4				
97	"	13-1	D*	RL 1	13	RL 1 coil
	"	13-2		"	14	
	"	13-3				
	"	13-4				
99	"	14-1	D*	RL 2	13	RL 2 coil
	"	14-2		"	14	
	"	14-3				
	"	14-4				

Fig. 4. Sheet showing that portion of TS2 which contains 10-1 and 10-2.

reveals that every pair shown is a return indication as evidenced by the asterisks.

Preparation of running sheets for the console is done in the same manner as for the rack. A rear-view sketch is made of the panel components, letters assigned to the terminal blocks and equipment and the running sheets made. In a console, the terminal block intended for making the external connections is usually the most common point for all circuits and may be used as a starting place for the running sheets. As the preparation of the running sheets progresses, cable runs are to be drawn on the console sketch.

Interconnection Sheets

Interconnection or cross-connection sheets show how to make the external connections between the rack and console, between the studio control room and master control and runs to all external units as loudspeakers, turntables, door lights, etc. Their preparation is similar to running sheets for racks and consoles and the same form is used. The sketch used to aid in their making is the preliminary conduit layout previously made. Instead of making reference to cable runs by letter as in the rack sketch, reference is now made to the conduit number in which the cables are to be run. To start, all terminal blocks involved and their terminals are listed on the forms. Then, with the aid of the block diagram, the "To" point of each "From" point is determined and written on the sheets. As with the running sheets, those pairs that are indicated as running back upon themselves should be so marked with an asterisk.

All that is necessary to complete the conduit drawing is to count the number of pairs shown on the interconnection sheets for each conduit and to list these quantities in the conduit table. The conduit drawing showing the points between which conduits are to be run and the conduit list that gives the individual conduit sizes, and the number and level of pairs each is to contain, gives the installer all the information needed to pull in all interconnecting cable. These cables, when connected in accordance with the cross connection sheets, completes the electrical part of the installation.

[To be Continued]

ULTRASONIC MAGNETIC RECORDING

• Characteristics and details of magnetic recorders essentially flat to 50,000 cps were revealed at a recent IRE section meeting in San Francisco. The speaker was Al Isberg, Chief Engineer of f-m station KRON, and during the war a project supervisor in the Airborne Instrument Laboratories on Long Island.

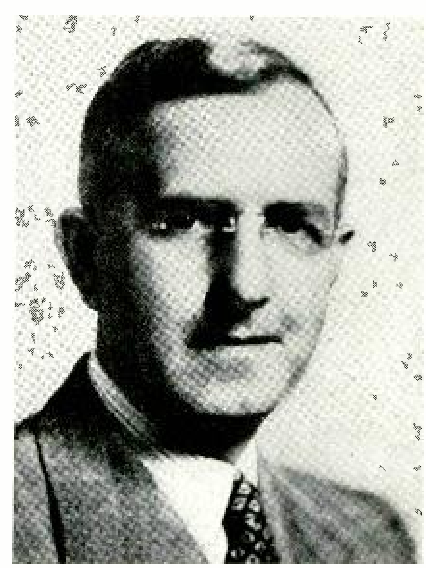
Security requirements permitted him only to say that the units described were used for the recording and later visual analysis of signal material of only a few seconds duration derived in part from radio receivers with i-f frequencies in the 5-mc region. Success of the project hinged on the fact that waveforms are not distorted when recorded in modulation envelopes, regardless of whether distortion in the recorder is independent of, or proportional to, frequency.

At the conclusion of this presentation, Mr. Isberg introduced Jack Mullen, San Francisco sound engineer, and Col. R. H. Ranger, who discussed the original discovery of Magnetophon tape recorders in Germany. Col. Ranger pointed out that military intelligence knew nothing of the recorder and its outstanding operational characteristics because the Germans had not labelled it secret. Mr. Mullen's "liberation" of several of the units called it to the attention of Col. Ranger, who returned to Germany again for further details and is currently active in the manufacture of both recorders and tape.

The meeting was concluded by demonstrations of a number of the available commercial wire and tape recorders.

NEW R.M.A. PRESIDENT

Max F. Balcom, Vice President and Treasurer of Sylvania Electric was recently elected president of the Radio Manufacturers' Association. He has been active in R.M.A. for the past twelve years, serving



Max F. Balcom

as vice president for two separate terms and as a board member for the past five years.

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

IT HAS BECOME increasingly apparent since the first introduction of vinylite records in the commercial field that somehow, some time, a compromise of sorts would be necessary between the shellac type of record and the pure vinylite. Unfortunately such a compromise is not easy to manage—the problem is a little like compromising between water and oil! From the listening point of view, however, it has become fairly clear that though the plastic record has the advantages of lightness, unbreakability, and lower hiss, it has numerous disadvantages as compared to the shellac, especially the high-quality laminated shellac disc. Most important from the listener's viewpoint is the type of noise that is heard on vinylite. One type, the tiny irregular click, though insignificant as a meter indication, turns out to be just as annoying, if not more so, than the even, steady hiss of good old-fashioned shellac. And worse, it can't be filtered out, since it extends to lower frequencies. Any irregular sound attracts the attention far more than a continuous one. An even worse type of plastic noise is the lop-sided hiss that comes from slight defects in material or pressing. Hardly a set is made without at least one such side. With nine-tenths of a record surface virtually silent, even a very faint hiss over a square inch or so of surface is extremely annoying—whereas the same hiss, uniformly distributed, would be quite harmless. Other difficulties that listeners experience with plastic are its amazing ability to attract dust particles (a wet cloth is the only solution), and its sensitivity to scratching. (Look at an "unbreakable" children's record after a week of play!)

Experiments in various plastics and and particularly in combinations of vinylite with other materials have been going on most energetically, and it is likely that very soon there may be some improved record materials on the retail market that in every sense approach a halfway point between the pure vinylite and the pre-war shellac. It looks as though only some such new development would make it possible for us to play records at high fidelity, "wide open," bringing through highs to ten or twelve thousand cps, yet with a minimum of the psychologically objectionable type of noise, and without fear of the quick wearing that spoils so many plastics when something less than the ideal pick-up is used.

Here are some recent recordings of outstanding interest technically:

Dvorak, Symphony #1, Opus 60. Cleveland Orchestra, Leinsdorf..... Columbia M687
A seldom heard but most enjoyable work by the composer of the familiar "New World" Symphony. Large orchestra, good resonance to hall. Wide-range recording with good surfaces. More of an "over-all" pickup than with Columbia's New York Philharmonic records, it seems.

Gershwin, Rhapsody in Blue (Arr.)
Selection of Hebrew Dances
Ambrose and His Orchestra (Embassy Club, London)..........................
London Decca, F5454, F5285
"ffrr" recording in a popular equivalent. This is British salon jazz, not particularly to the U. S. taste. Quite wide-range recording, more liveness than in most American populars, but considerably less than the big classical ffrrs. (Two samples from a larger series.)

Debussy, Sonata #3 for violin and piano (1917) Zino Francescatti, Robert Casadesus............... Columbia X280
A new wider range recording with excellent liveness, fine balance between violin and piano. It supplements older version (pre-war) by Szigeti and Foldes, Columbia X242. Interesting comparison technically. The old set is minus highs, violin is more prominent, closer. But over-all liveness is excellent, too.

Purcell, Abdelazer Suite. Vox Chamber Orch. Edvard Fendler.......... Vox 199
Numerous recordings are still being issued with the pre-war type of restricted range—nothing above 5-6000. This is good example of it—small string orchestra, excellent liveness, good recording. But no highs in the record, and surface is poor. Sounds fine on standard console phonographs!

Liszt, Mephisto Waltz. N. Y. Philharmonic, Rodzinski....... Columbia X281
Apparently this was made before recent improvements in Columbia's Philharmonic recording technique. Wide range, but upper part of "curve" seems lower—less rising characteristic than in later issues. Also liveness is different; more over-all effect here, apparently less accentuation. Sounds a bit tinny, "narrow." Turnover point is evidently higher than on newer Columbias, making for less positive bass. Compare with Kahchaturian, Gayne Suite or Rimsky-Korsakoff, Russian Easter Overture.

Sir Lancelot, Calypso Carnival. Trinidad Serenaders; The Volunteers...............
Crest Recordings CT-3
Excellent highs for this type recording; snare drum, etc. come through clearly, also sibilants in vocal. Heavy recording, tends to "buzz" a bit.

Shostakovitch, Symphony #9.
a) Boston Symphony Orchestra, Koussevitsky....... Victor M1134
b) New York Philharmonic, Efrem Kurz............. Columbia M688

The Columbia set has the better surfaces, and frequency response appears to extend to the 10,000-cps range. Doubtful if Victor set exceeds roughly 7,000 cps. Columbia recording has greater breadth of sound than Victor, showing a legitimate area of difference in technique; both use overall type of pick-up, but Columbia uses more accentuation. This is clearly an arguable point. and constitutes an interesting comparison. (Both sound equally well on average phonographs.)

MUSICAL ACOUSTICS

BENJAMIN F. TILLSON*

PART IV

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

THE APPRECIATION of music may result from either or both of two approaches: the emotional and the intellectual. The former is not subject to a definite analysis as it depends upon the personality of the individual, his environment, his mental serenity, health, disposition, associated ideas, heritage, education, experiences, and still other factors.

Although we do not know why, most of us probably do not find the same appeal in a melody and a harmony, but particularly the latter, as it is transposed in different keys. There have been elaborate discussions concerning the emotional and esthetic suitability of each key. It is not likely that all persons would hold the same opinions concerning the expressiveness of each key.

Someone might question whether such descriptive variants in musical keys were not due to the tempered scale causing in the different keys a different sequence of intervals divergent from "just intonation."

But, playing a phonograph record at different speeds will maintain the original interval-frequency ratios and will change the keys. A listener may thus determine for himself whether the theme and harmony holds for him the same emotional appeal in all keys.

A composer selects some certain key or keys as the best vehicle for the musical expression of his message or poem. Most lovers of classical music fail to realize that they do not hear it reproduced today as the composer intended it to be heard. The conductors, arrangers, and publishers fail to make allowances for the changes of pitch which have occurred in the years since composition. Then, too, the timbre of musical instruments has also changed.

Vocal music of over 150 years ago must be transposed a whole tone lower than present standard pitches in order to duplicate the emotional sound effects for which it was then composed.

The vocal music of Handel's time, the music of Mozart, and probably that of Haydn and Beethoven, should correspondingly be transposed one-half tone lower, because "C" then varied from 500 to 513 cycles per second. Handel's tuning fork of 1751 gave a "C" of 510 cycles/sec. The pitches were different in Germany, France, Italy, England, and Belgium, affecting the operas therein; and during 130 years the pitch of "C" rose from 467 to 546 cycles/sec. There was a lack of standardization in either the pitches, the laying of the temperament of the scale in tuning the scale intervals of instruments, and in the timbre of instruments during the golden age of classical music. So there was not adequately uniform ear training in music.

Since then, during the past fifty years, through the medium of the phonograph and radio, the public's listening opportunities have been intensified and extended more than heretofore in the realm of music. But the defective fidelity and distortions of such music programs have trained ears to become accustomed to them, and have destroyed a finer appreciation.

Listener Preference

Most listener reaction surveys in recent years have concluded that the public did not wish to hear the higher frequencies; and even that the violins of the old masters were superior because they lacked the high frequencies. However, some such tests have not determined and reported the complete technical data necessary to make an intelligent analysis of their conclusions; nor have they avoided the psychological pitfalls which are possible in the statistical sampling of human reactions, opinions, and experiences.

Much more is involved in music than the qualitative and quantitative analysis of its component tones and their harmonics. The duration of time for each harmonic, or conversely its damping, makes a great difference in musical pleasure. With the damper pedal of the piano one can hear how displeasing is the jumble of sustained tones from a piano.

The merit of the violins of the masters probably rested in their uniform resonant response and a lack of extreme damping of tones, and lack of harmonic modulations. Such aberrations are predominant in the higher frequencies. With imperfect instruments and reproduction, relief from them is most easily attained by doing away with the higher frequencies; and the public would naturally prefer such devitalized music rather than distressful distortions.

The best of lateral-cut commercial phonograph records or radio transmitters rarely give undistorted frequencies above 5,000 cycles per second and often not better than 3,000 cycles with the loud speakers commonly supplied. A frequency band two octaves wider (up to 11,000 cycles) is claimed for British hill-and-dale disc recording and Western Electric Co. recording of vertical-cut records. Sound films have a range to 9,000 cycles, as has also been claimed for magnetic recordings, which more frequently are below 5,000 cycles.

AM radio transmitters do not exceed 7,500 cycles because of the limitation in the width of licensed wave bands to avoid crowding the commercial radio spectrum. FM radio transmitters are required to be adequate for a range of 50 to 15,000 cycles; but lack of financial resources, coupled with the restrictions of musician unions, rarely permit programs of such frequency range because the radio stations are forced to use phonograph records and telephone lines limited to 5,000 cycles, or possibly 8,000 cycles if the increased cost can be afforded.

Most a-m radio programs also pass over telephone lines from the studio to the transmitter with the same restriction of 8,000 cycles. Although the telephone companies say that they can offer a service range up to 15,000 cycles at a premium cost, the source of this information thought that no one was leasing such high-frequency service.

Musical Education

Education should promote an appreciation of the finer things in life. That requires training. The most profitable period for training is in childhood, before habits have been set. Music appreciation has been accepted as one of those finer things proper in public school education; but, while its content has been given attention its quality has been too greatly ignored. Ear training in the schools offers the best method for counteracting such a low grade aural environment as has been thrust upon our youth by juke boxes, phonographs, and radios.

During the first century A.D., Plutarch stated in his "Opera Moralia": "Whoever it be that shall give his mind to the study of music in his youth, he will be sure to applaud and embrace that which is noble and generous, and to rebuke and blame the contrary, as well in other things as in what belongs to music. And by that means he will become clear from all reproachful actions, for now having reaped the noblest fruits of music, he may be of great use, not only to himself. but

Consulting Engineer, Montclair, N. J.

to the commonwealth; while music teaches him to abstain from everything that is indecent, both in word and deed, and to observe decorum, temperance, and regularity."

However, times change; and if Plutarch heard some of the indecent songs now "plugged" on the radio under the guise of music his idealism might have been tempered by the realization that such tools of great potential service to mankind may be prostituted to evilly destructive uses.

There exists to the reproducers of music and the makers of reproduction devices a great opportunity and challenge for public service, as well as commercial profit, which they have failed to realize to the full during the past sixty years since the phonograph was developed.

By greater attention to the higher musical fidelity of their products and performances, and by more earnest cooperation in cultivating the finer appreciation of music by our youth, they will open markets of vaster importance than their present conception. They will halt the dying listener interest in phonograph and radio music. It seems irrational for them to believe their reports that listeners prefer not to have high fidelity. If that were so they and others would avoid the personal performances of musicians and orchestral concerts.

Perhaps, as in other fields, there is a need for an Institute of Sound-Reproducers to integrate an approach to the public relations and research problems of their industry, with a staff free from the inhibitions of past habits and experiences in the phonograph and broadcasting fields, and guided by one with technical imagination and broad interests.

The Acoustic Problem

Like most important projects, the problem is not simple. But is it receiving the attention and effort it warrants? Or has the tendency been to avoid the problem by rationalizing a conviction that it does not merit solution, because the vast majority does not want what it has had no adequate opportunity to learn to appreciate—high fidelity.

There would have been no progress in the world if such an attitude had prevailed in other fields. Most new developments have been opposed by the majority, even the advent of the Machine Age. Public education, itself, is still compulsory.

The development of the science of electronics was probably the greatest incentive in much-needed research for improvement in the acoustics of phonographs. But its apparent magic has, perhaps, its drawbacks if it encourages the *attitude* that fundamental, primary faults can be compensated for and alleviated electronically; and, therefore, there is no incentive to rectify the sources of trouble.

The phonograph and radio arts need to develop a true reproduction of live programs, and then any further circuits and gadgets can be added to permit the listener the self-expression of altering the true to suit his fancy and musical ignorance.

It is probably generally true that whenever harmonic electric oscillations are subjected to electric or magnetic fields or impedances, whether capacitive or inductive, they have the hazard of distortion. The more such means are employed to correct aberrations the greater the opportunity to lose certain attributes of fidelity, liveness, and that naturalness whose lack promotes psychological fatigue. The deficiency may be only in transients which the usual methods of testing fail to reveal in average or root-mean-square observations.

Unfavorable listener-reaction studies of high fidelity programs may be due to such transient or intermodulation deficiencies in true quality as were not appreciated by the listeners or those who conducted the tests.

Of all the branches of physics, sound appears to have been most neglected in college science courses; and it is generally ignored by musicians and architects.

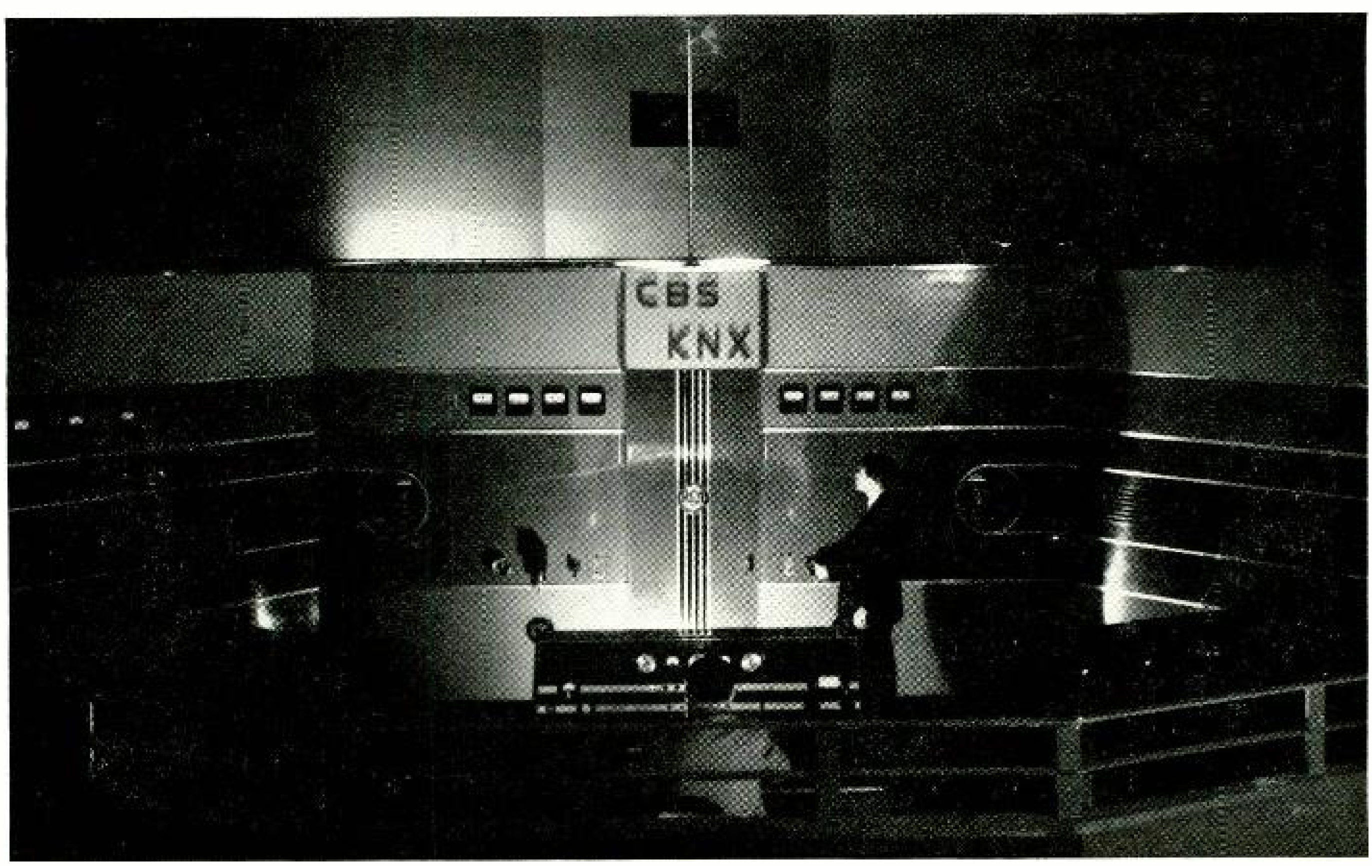
Causes of Distortion

The following classification of distortions accords with the effects produced:

1. External noises or sounds which affect the electro-mechanical systems for detection, transmission, or reproduction of sound programs. These chiefly occur from microphonic vibrations of electron tubes without shock-mounting or acoustic insulation from the loudspeaker output.

[*Continued on page 43*]

Operator at the center control console of CBS-KNX.



Improved Method of
Locating Operating Point

ARTHUR PLOWRIGHT

VARIOUS TECHNIQUES which may be utilized to find the operating point for the circuit of *Fig. 1* have been presented during the past few years.* All assume constant gain or require plotting of curves. The improved method to be described necessitates only a draftsman's scale or machinist's rule, involves no assumptions of linearity, and requires no construction lines.

Constant Gain Technique

A method in which constant gain is assumed is shown in *Fig. 2.* A load line is drawn with a slope equal to the negative reciprocal of $(R_L + R_k)$. At its intersection with $e_c = 0$, a perpendicular is dropped to the point A'. Then a somewhat lower value of plate current than that at $e_c = 0$ is chosen and its drop across R_k is computed. This point is located at B (for $i_p = 50$ ma), and a perpendicular is erected to intersect with this value of current at B'. A straight line joining $A'B'$ intersects the load line at the approximate operating point. The point thus located is not quite accurate because of the underlying assumption of uniform gain.

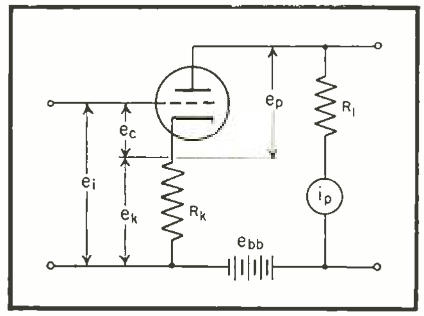


Fig. 1. Conventional schematic diagram including information usually supplied.

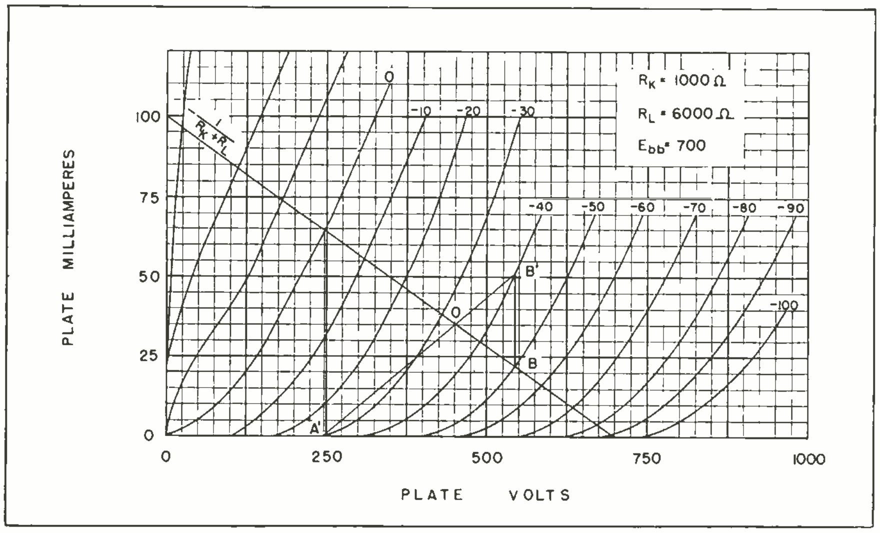


Fig. 2. Assumption of constant gain incurs certain error. Constructions are required.

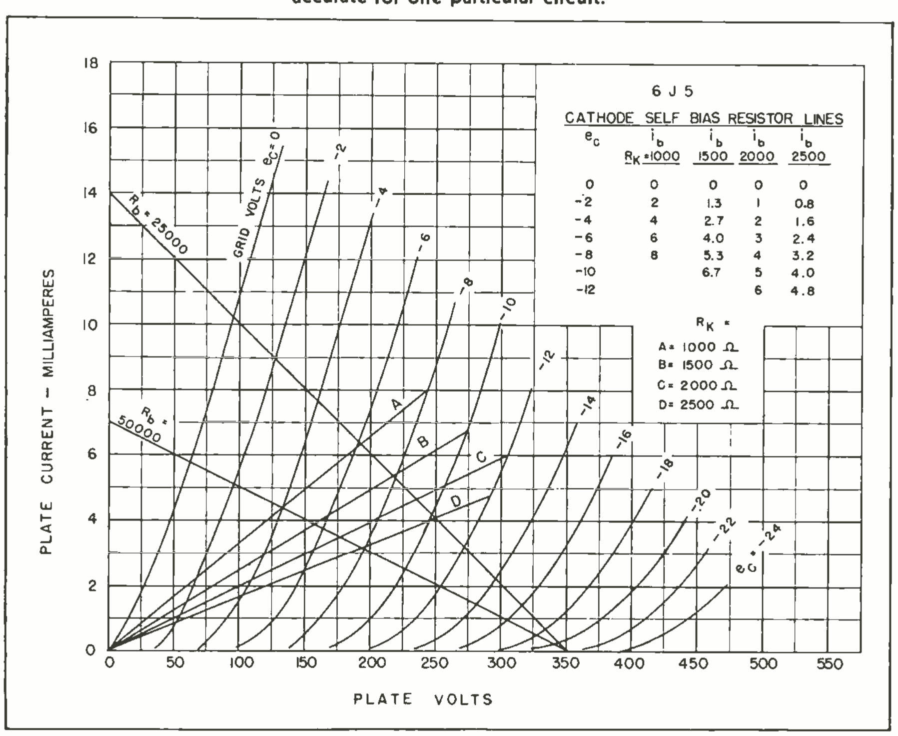
Cathode Characteristic Technique

A method which eliminates the foregoing inaccuracy, but which requires plotting of points and curve construction is shown in *Fig. 3.* Here the drop across R_k is plotted in an e_c-i_p frame of coordinates, so that the intersection of the cathode characteristic with the load line may be used as a criterion of equilibrium and thereby a specification of the operating point.

This method is capable of giving accurate determinations provided it is applied to a specific situation. In this case, the slope of the load line is chosen as for the first demonstration and the location of the operating point involves no approximations. In *Fig. 3*, curves are drawn to present a general situation involving a range of cathode resistance and load resistance values. Since the cathode resistance cannot be conveniently included in the load line, determinations on this base are correspondingly inaccurate.

It must not be overlooked, however, that when applied to a specific circuit

Fig. 3. Cathode characteristic can be graphed, using e_c-i_p coordinate frame, and will be accurate for one particular circuit.



*Arthur Shach, *Electronics*, Feb. 1944
Paul Hunter, *Electronic Industries*, Nov. 1945
John F. Rider, "Inside the Vacuum Tube," 1946

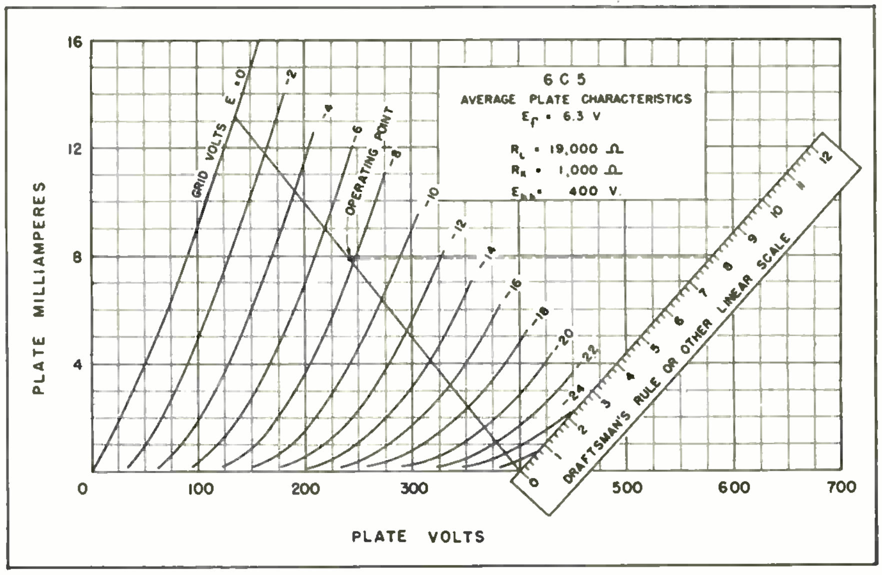


Fig. 4. Improved method of locating operating point involves only the application of an arbitrary linear scale.

with R_k included in the load line, that the method is capable of precision. Note that *A*, *B*, *C*, and *D* are slightly concave upward rather than straight lines, as required by the coordinate reference frame.

Direct Determination Technique

The new method next described requires the computation of one point and the application of a linear scale, such as a draftsman's rule. In the diagram of *Fig. 4*, the load line for the circuit is drawn as before, for the circuit of *Fig. 1*. The cathode resistance is 1000 ohms, and the drop for 10 ma, for example, will be 10 volts; the "10" on the rule is placed at 10 ma, while the zero of the rule is placed at a convenient point along the e_p axis.

Since the condition of equilibrium of the circuit is that $e_c = e_k$, the operating point is now found by matching values of e_c along the load line with values of e_k along the rule. This may be done running the eye along the horizontal coordinates, or a sheet of paper may be slid down the chart until matching values are seen along the edge. For the case illustrated, this condition evidently takes place at $e_c = -7.9$ volts and the problem is solved.

Specified Grid Bias

One of the most difficult procedures in non-linear analyses is to determine desired operating conditions. We wish to find, for example, the most favorable value of cathode resistance for the situation of *Fig. 4*. Using techniques of plotting the cathode bias characteristic as shown in *Fig. 3*, the labor is multiplied as it becomes necessary to make several plots (preferably for the RMA values) and then to select that which most nearly meets the desired operating condition.

This selection may be made readily with the new technique as follows: Let it be specified that R_k shall place the operating point as near to $e_c = -10$ as practicable. This means that e_k must likewise be equal to 10 for equilibrium. It is usually desired to maintain R_L at its original value, changing only R_k to meet the bias requirement.

In the event, however, that the sum of R_L and R_k may be maintained constant, it is only necessary to note that a current of 6.4 ma flows at the intersection of the load line with $e_c = -10$, or the cathode resistance is 1562 ohms correspondingly. The nearest 10% or 20% RMA value is

[*Continued on page 39*]

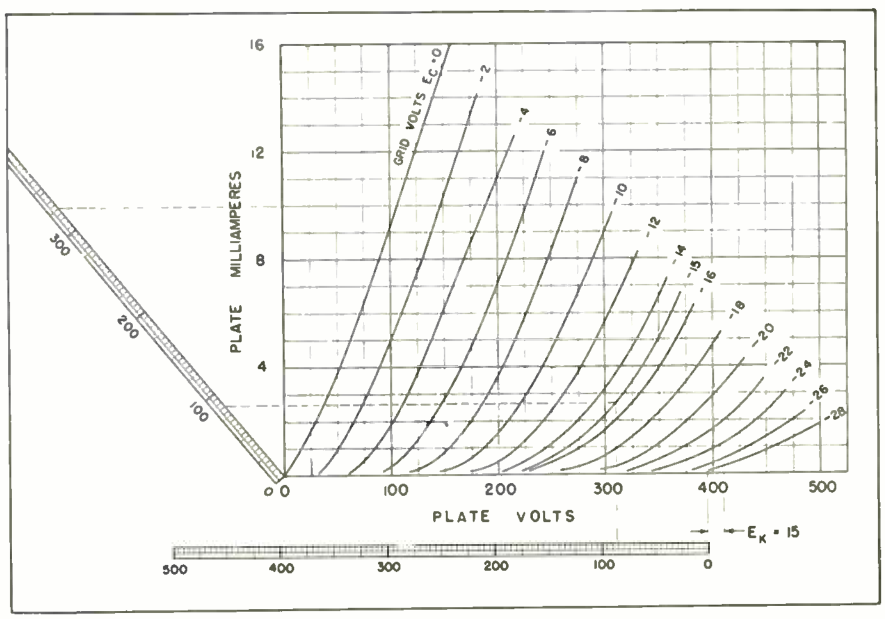


Fig. 5 (Center). When R_L is constant, and operation is desired at a specified grid bias, R_k may be found as illustrated.

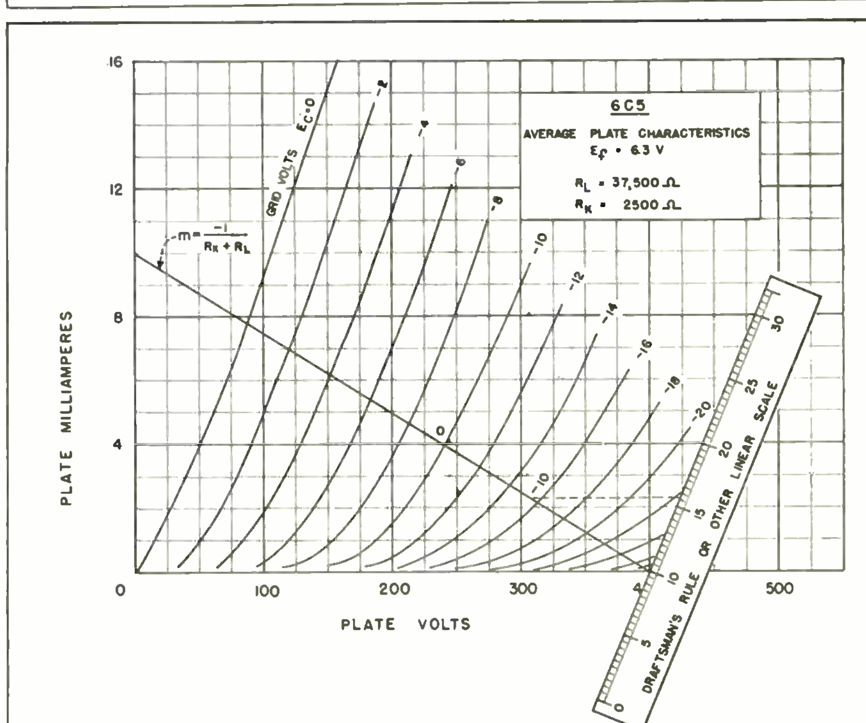
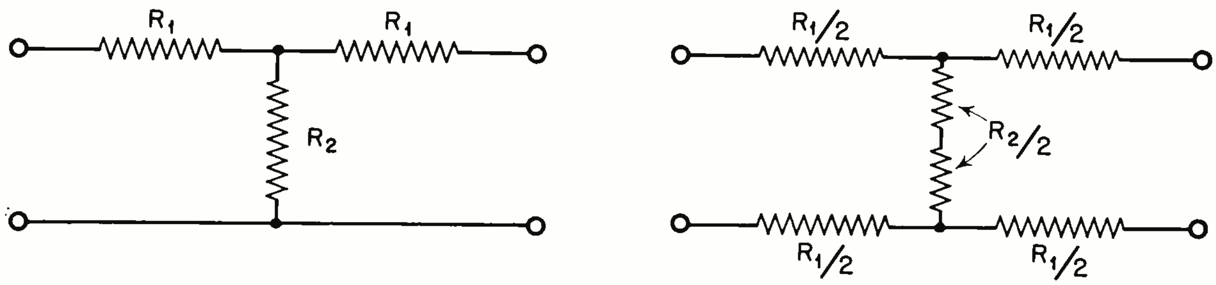


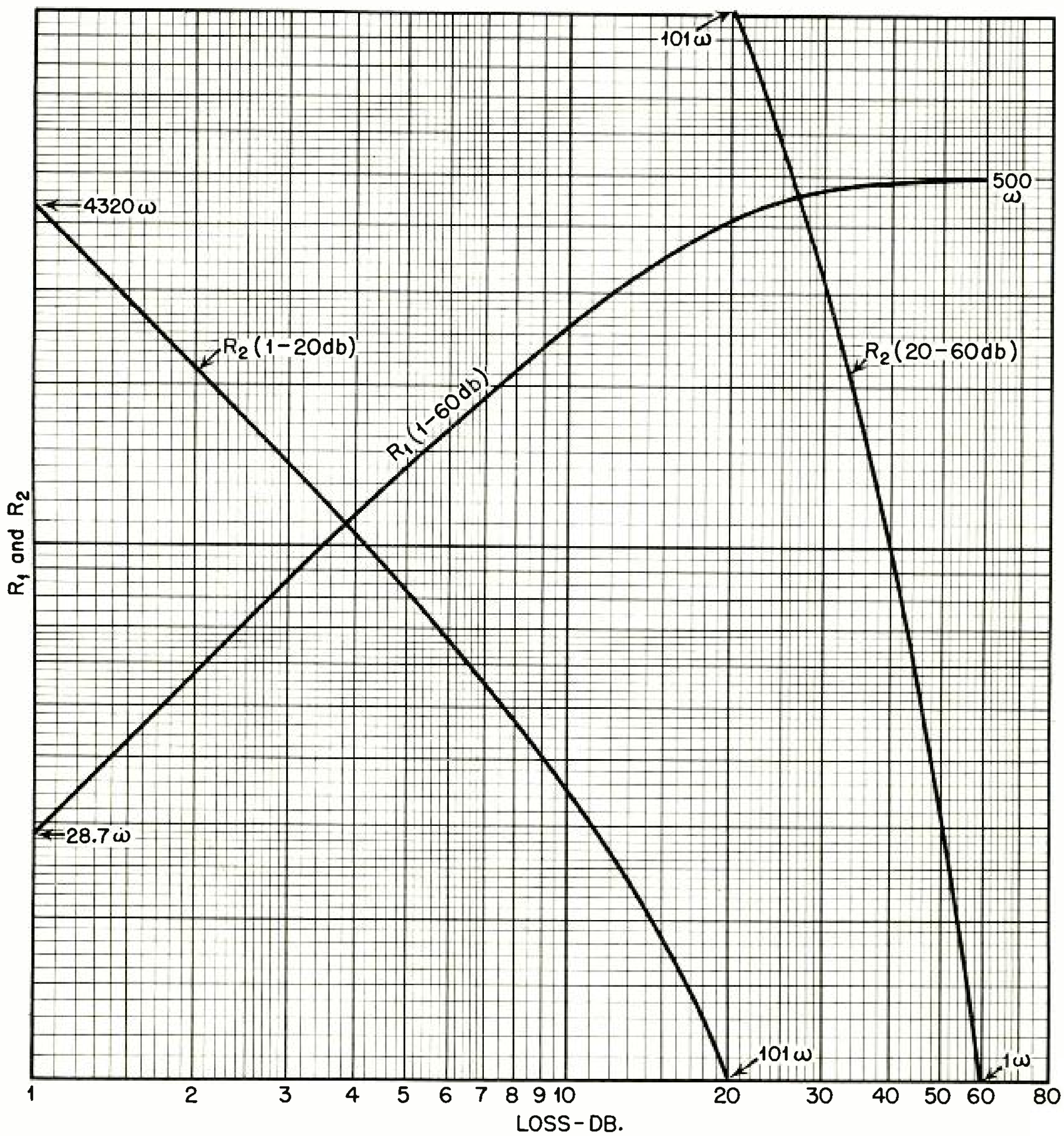
Fig. 6. Finding operating point when R_L = 37,500 and R_k = 2500 ohms.

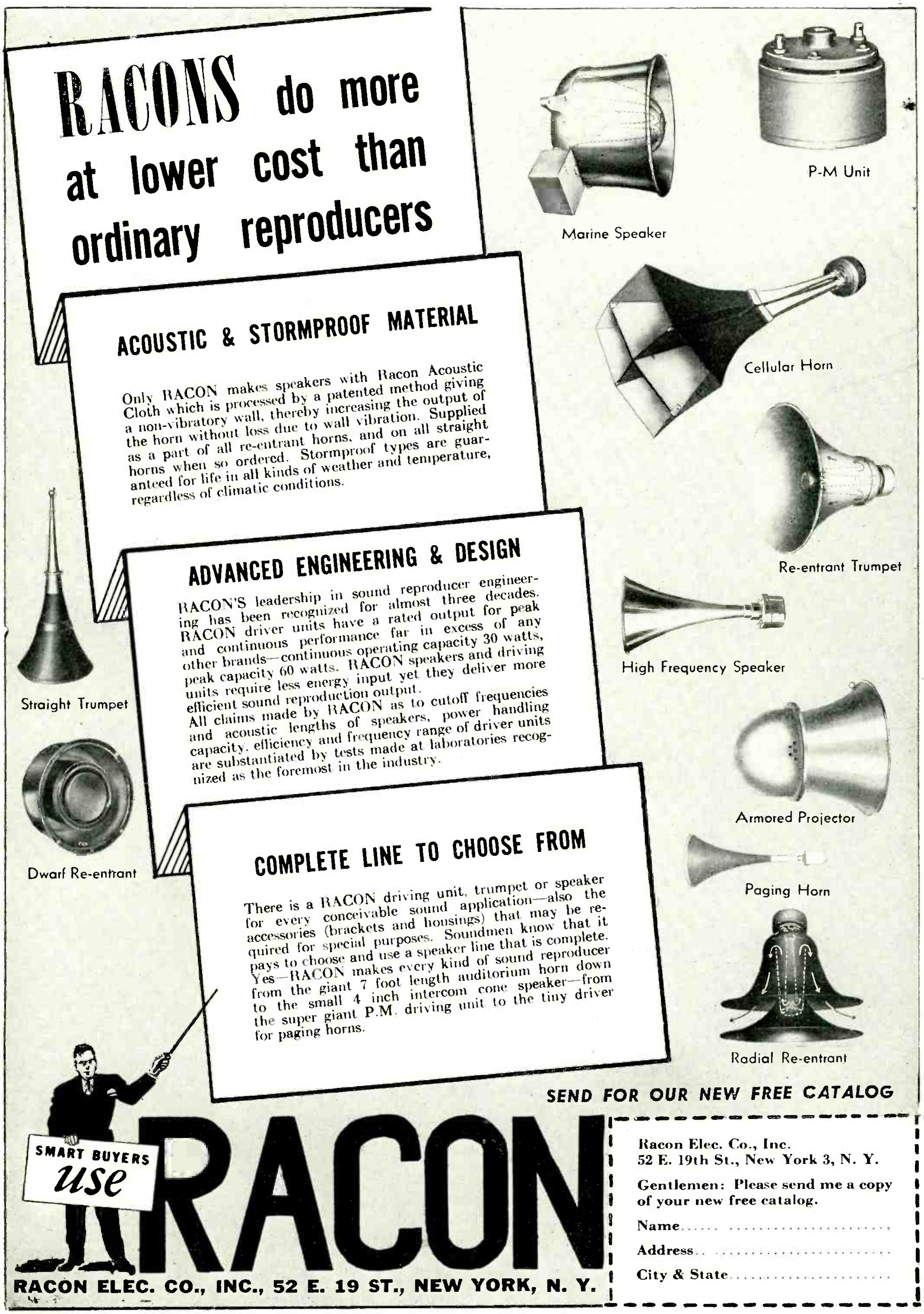
AUDIO DESIGN NOTES

DATA FOR MATCHING PADS WITH MINIMUM LOSS



FOR ANY OTHER LINE IMPEDANCE MULTIPLY R_1 and R_2 BY $Z_0/500$

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Only RACON makes speakers with Racon Acoustic Cloth which is processed by a patented method giving a non-vibratory wall, thereby increasing the output of the horn without loss due to wall vibration. Supplied as a part of all re-entrant horns, and on all straight horns when so ordered. Stormproof types are guaranteed for life in all kinds of weather and temperature, regardless of climatic conditions.
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P-M Unit
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TECHNICANA

FM PILOTUNER

• While not strictly within the field of audio, this tuner provides a means of obtaining an audio signal from FM signals, as an element in a high-fidelity radio receiver system.

The Pilotuner is a compact FM tuner, employing only five tubes and a selenium rectifier. The circuit consists of a tuned r-f stage, a combination first detector-oscillator, two stages of i-f amplification, and a ratio detector. It is ruggedly built on a 6 x 8 inch chassis, and enclosed in a wood cabinet. The tuner has a sensitivity of approximately 25 microvolts for an output signal of 0.1 volts, which is sufficient for most audio amplifiers. The radio detector provides perfect quieting, and the audio response is extended up to 12,000 cps, with the circuit being designed with somewhat less de-emphasis than is required by transmitter standards. This gives some equalization to compensate for drooping high-frequency response in existing AM receivers.

One feature of the Pilotuner circuit, *Fig. 1*, is the inclusion of two r-f chokes in the leads to the power transformer, and the 50-μμf capacitor for coupling the power line to the antenna coil. This arrangement eliminates the need for an outside antenna when the tuner is used in areas having a fair signal strength—and it is, in fact, perfectly satisfactory in New York City where interference and building absorption is particularly pronounced.

The tuner is equipped with a combination power switch and signal transfer switch, so that the output of the tuner is connected to the phonograph input terminals of an AM receiver and the phonograph pickup leads are connected to the tuner. When the tuner power switch is actuated, the output is fed to the receiver through the shielded lead, with the phono switch on the AM receiver in the phonograph position. When the tuner is turned off, the output of the phonograph pickup is fed to the receiver.

The half-wave selenium rectifier circuit is followed by a resistance capacitance filter which effectively precludes any trace of hum in the tuner output.

CATHODE-FOLLOWER PRECAUTIONS

• Users of miniature tubes are cautioned against employing the 6AG5 and 6AK5 types in cathode-follower circuits, since

[*Continued on page 46*]

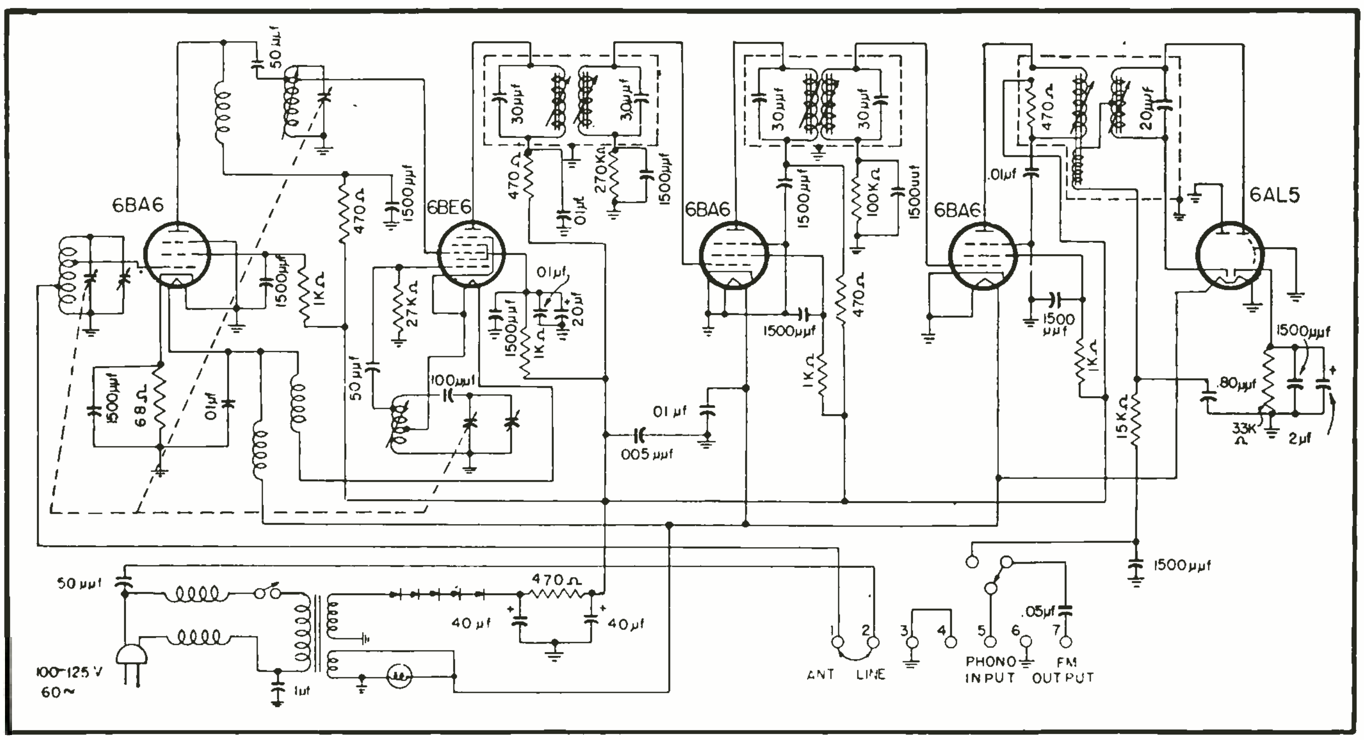
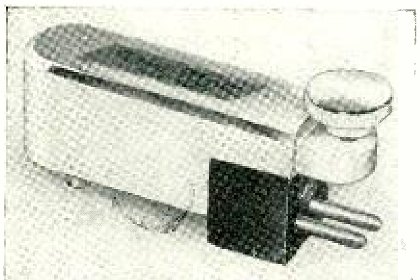


Fig. 1. Schematic diagram of the Pilotuner

NEW PRODUCTS

PICKUP ADAPTER

Technical Products International, 453 West 47th St., New York 19, N. Y., has recently announced the Vibromaster Type M adapter, designed to permit the use of



the new Pickering Cartridge or the General Electric Variable Reluctance Pickup with the standard Western Electric 5A arm.

VERTICAL REPRODUCER CARTRIDGE

Pickering & Co. Inc., 29 W. 57th St., New York 19, N. Y., has recently announced the new Model 131 Vertical Cartridge, identical in size, appearance, and mounting, with the lateral cartridge.

The vertical unit has an output of approximately 20 millivolts, and is designed to work into a grid circuit, with the necessary equalization furnished by the preamplifier. The frequency response is flat on a velocity basis, and excellent response is obtained to upward of 10,000 cps. As with all Pickering Pickups and Cartridges, the intermodulation and harmonic distortion are held to a minimum.

Tracking pressure for the vertical cartridge is approximately 1 oz. This value may appear to be high in the light of modern lateral pickups, but considerably greater pressure is required with vertical reproducers in order that the entire up and down motion of the stylus be translated into effective movement of the magnetic structures. The pickup arm should show no tendency to follow the vertical modulation on the record.

The Model 131M Cartridge is supplied with a 0.002" sapphire, with diamond styli being available at an extra charge.

D-C AMPLIFIER

Specifically developed to accommodate the General Electric variable reluctance magnetic pickup without the use of any additional preamplification or equalized circuits.

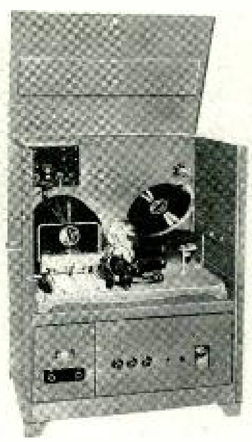
Equipped with a built-in specially designed low-noise and low-hum pre-amplifier and fixed pre-equalizer to fully compensate for the characteristics of the G. E. variable reluctance pickup. It contains, in addition, a variable high-frequency equalizer for compensation of pre-emphasized recorded and radio programs, as well as a low frequency equalizer for full compensation of constant amplitude recordings.

Filtered dc, having less than .03% ripple, is applied to the heaters of the input tubes through a regulating ballast resistor to stabilize heater voltage and insure minimum heater to cathode leakage.

Utilizes a new signal self-balancing and current drift-correcting direct-coupled output circuit. Response is 20 to 20,000 cycles $\pm$1 db. Develops 23 watts with less than 1% total distortion. Less than ½ of 1% is present at a 12 watt level. Overall gain; 117 db. Hum and noise level;—40vu. An additional independent input of 500,000 ohms is provided. Balanced output terminals are provided for 8/16/20 and 500 Ohms. In-between terminals provide the following additional output impedances; 2/4/5/10/80/125/160 and 175 ohms.

Entire input section is mounted on a floating plate.

Manufactured by the Amplifier Corp. of America, 398 Broadway, N. Y. 13, N. Y.



ROBOMAT

Eastern Amplifier announces production of a completely automatic sound reproducing system for industrial organizations, stores, offices, restaurants, ships, churches, cemeteries, mortuaries, skating rinks, country clubs, swimming pools, etc. The new product is called the "Robomat."

This unit can be used to replace programmed or continuous wired music installations. "Robomat" includes a microphone for instantaneous paging. A built-in powerful AM radio receiver for special events gives excellent results even on stations with a weak signal.

The record changer is heavy duty and automatic. It stacks 20 records and plays both sides or a total of 40 discs. This record changer can be operated for continuous periods on either a planned program, such as one record every three minutes, or on a continuously operating program. All you do is flip a switch. The "Robomat" is substantially shock mounted so that it can operate under extreme vibration conditions, such as encountered on ships, trains, etc.

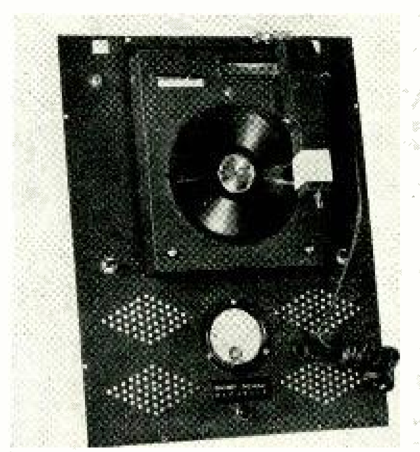
The output capacity is 90 watts to assure coverage of 50,000 square feet in noisy indoor areas, 200,000 square feet for background purposes and a one-quarter mile radius outdoors when used with proper speakers. The "Robomat" goes into operation automatically in the morning and shuts down automatically at night! It requires no attention other than to change records once a day. If records are not changed it merely repeats the same records the following day.

WAGNER-NICHOLS RECORDER

In response to many requests, the Wagner-Nichols Recorder Mfg. Co., 67 W. 44th St., New York 19, N. Y., has set aside a limited number of recording units for those interested in assembling a complete recorder using their own amplifier and speaker equipment. Recent improvements on the original design have resulted in a reliable unit employing a unique lead-screw arrangement which records at 416 lines per inch. Fifteen minutes of program time may be recorded on each side of a single 4⅜" Vinylite blank.

These basic units are available complete with 60-cps, shock-mounted 33⅓rpm motor free of wows or vibration, recorder chassis and mounting plate, crystal embossing and reproducing head, and a supply of blank disks. Full instructions regarding their use will be enclosed with each unit.

Less than one watt of audio power is required to drive the embossing head, and the output level from the playback is approximately 0.25 volts. For embossing, the head is coupled to the output plate of an amplifier through a capacitor, while conventional circuits are used for the playback connection. A complete recording channel may thus be assembled with the addition of an amplifier, speaker, microphone, and switching facilities. For recording radio programs, the connections may be made to the output stage of any



receiver capable of supplying the necessary output voltage. An important innovation in this device is the fact that the same stylus and head are used for both recording and reproduction.

This recorder, first announced in the May issue of this magazine, has been improved considerably since its first showing, and is now ready for distribution to technically inclined users. The complete recorder, including amplifier, microphone, and speaker, all mounted in an attractive case, will be available in a few months.

[Continued on page 48]

Volume Expander

[*from page 9*]

trol. It is merely necessary to substitute for the control signal a 22½-volt battery and potentiometer. The volume can then be varied smoothly over a range of about 15 db, and the connecting cables can be of any length whatever, since they carry only direct current.

A little care must be exercised in using the volume expander. It should never be used on program material where the source of sound is inherently incapable of a volume range of more than 20 to 30 db. This applies to solo instruments (other than piano and organ), solo voices, string quartets, and so on. On orchestral, choral, and organ music it can be used on almost any record with excellent effect. The actual manner in which the original recording was controlled determines whether 8 db or 12 db of expansion can be used. Paradoxically enough, the wider the volume range on the original recording, the more expansion can be tolerated.

In *Fig. 4*, it will be noticed that in the 12 db position the input signal necessary to cover the entire range of expansion is about 29 db. This is about the volume range of a good modern recording. When playing such a recording it is best to set the expansion control so that the eye of the indicator tube fully closes on average peak levels. The expansion will then be completely off on very low-level signals. On records of more restricted range, it is best to set the expansion control so that surface noise just does *not* operate the indicator. This will then give the maximum increase of volume range.

It is good practice to install the expander with a gain control following the unit. The expansion control can then be left in the full-on position, and the input gain control used to adjust for input signal peaks. The output gain control then controls loudspeaker volume, and all of the output peaks will be at the same level, regardless of the actual level on the recording.

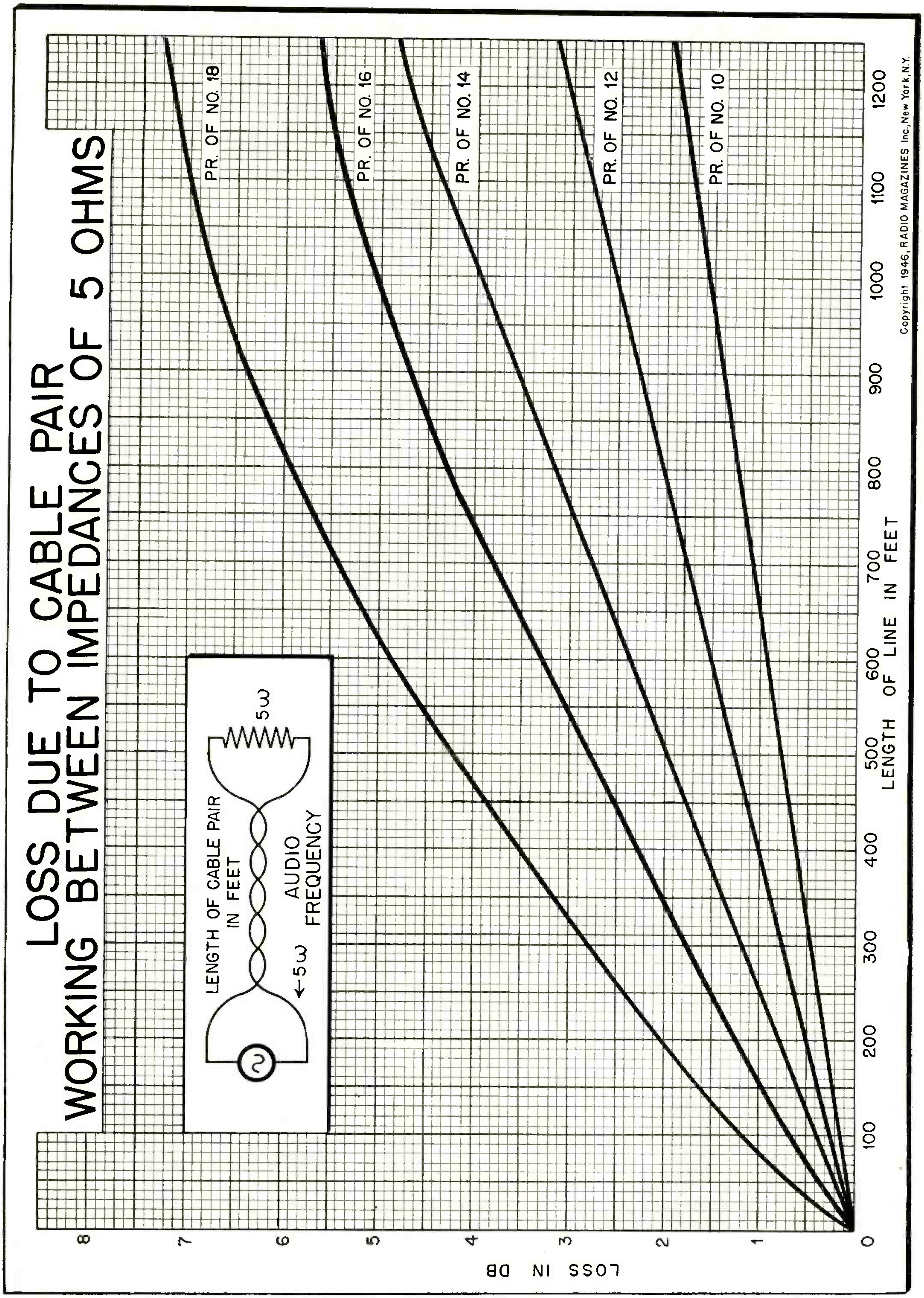
Operating Point

[*from page 33*]

1500 ohms, which would therefore be chosen, leaving 18,500 ohms for R_L, for which the nearest 10% value is 18,000 ohms. The result obtained is precise, even if the limited range of commercial resistance values prevents use of the exact value.

Returning to the condition that R_L remain constant, we may make the following summary of a particular problem: R_L = 32,500 ohms, R_k is unknown, E_{bb} = 410 volts, e_c = −15. Required to

[*Continued on page 41*]



LOSS DUE TO CABLE PAIR
WORKING BETWEEN IMPEDANCES OF 5 OHMS
PR. OF NO. 18
PR. OF NO. 16
PR. OF NO. 14
PR. OF NO. 12
PR. OF NO. 10
LENGTH OF CABLE PAIR IN FEET
AUDIO FREQUENCY
5ω
←5ω
LOSS IN DB
0
1
2
3
4
5
6
7
8
100
200
300
400
500
600
700
800
900
1000
1100
1200
LENGTH OF LINE IN FEET
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[*Continued from page 39*]

determine the necessary value of R_k precisely. Since the final slope of the load line is likewise unknown, we must start with the available locus, which is that of $e_c = -15$. It is desired to travel down this locus to the point of equilibrium. When this point is found, i_p will be known as well as its drops across R_k and R_L.

The condition of equilibrium is that $E_{bb} = e_p + e_L + e_k$, and that $e_k = e_c$. This latter fact allows a solution, since we may write $E_{bb} = e_p + e_L + e_c$, or $E_{bb} = e_p + e_L + 15$. Since $e_L = i_p R_L$, a linear scale may be placed across the coordinates as shown in in *Fig. 5*. Writing $i_p R_L = E_{bb} - 15 - e_p$, another linear scale with intervals equal to those of the plate volts axis may be placed to graphically subtract $15 + e_p$ from E_{bb}, also shown in *Fig. 5*. All figures read on this reversed scale are possible values for the right side of the foregoing equation; all figures on the oblique scale are possible values for the left side of the foregoing equation.

To find the equilibrium condition, the left and right sides of the equation must be equal. Therefore, slide a draftsman's triangle or a sheet of paper with its corner on $e_c = -15$, keeping the right-angled device square with the coordinates as shown in *Fig. 5*, until the same value is read on both scales.

This evidently occurs at $i_p = 2.75$ ma. To obtain 15 volts drop at 2.75 ma obviously specifies that $R_k = 5450$ ohms, and the problem is solved precisely.

Unbypassed Cathode Resistor

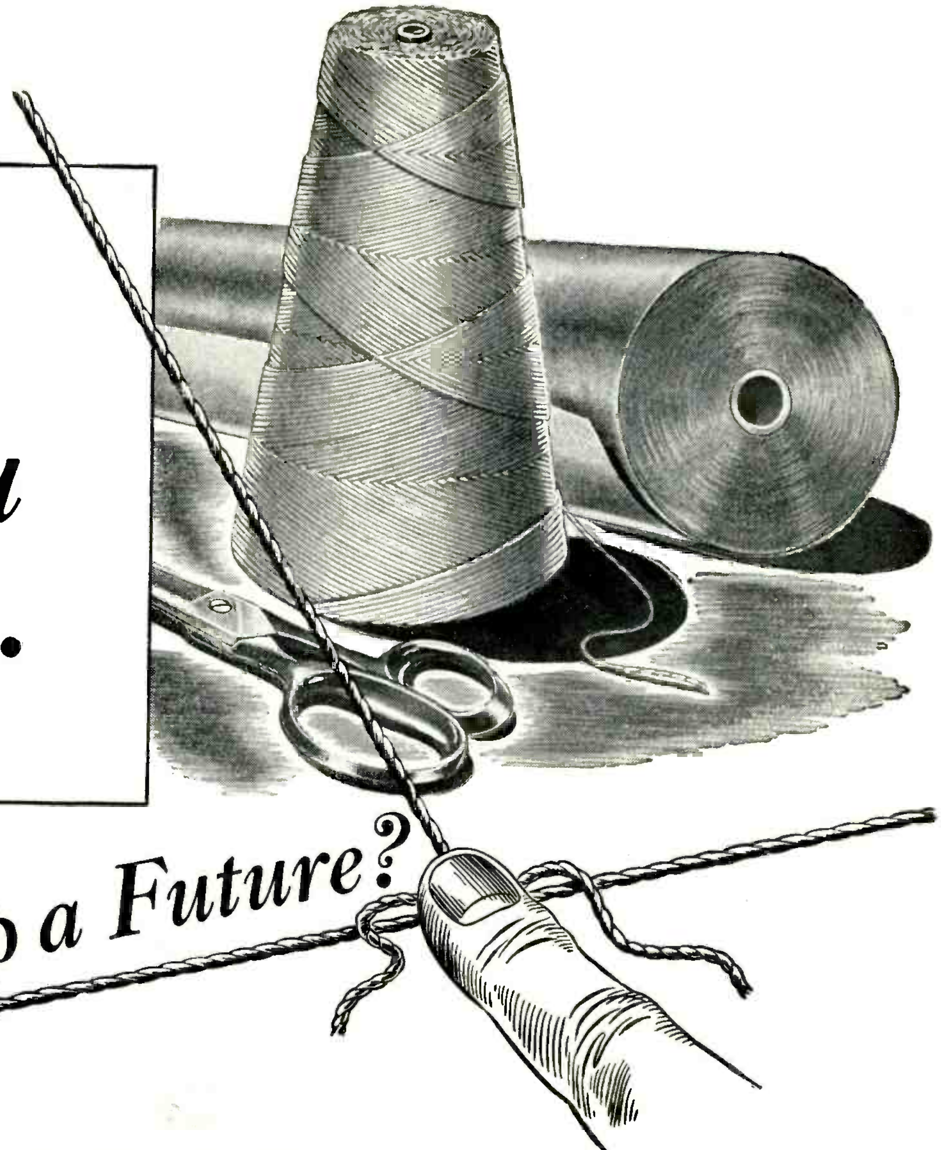
This technique is further adapted to finding integral values of input voltage along the load line for R_k, unbypassed (see *Fig. 1*). In this event, an a-c component appears across R_k which is of opposite polarity to e_t. We observe that at any time, $e_c = e_t - e_k$; when $e_t = 0$, $e_c = -e_k$, which relation was utilized to locate the operating point.

With current feedback, and $e_t = -1$, $e_c = -1 - e_k$, or the rule in *Fig. 4* may be slid parallel with itself to make "1" fall on the e_p axis instead of "0". Then the point of equilibrium is found for equal values of e_c and e_t, as before. But since the scale for e_k has been moved one volt, the input voltage is precisely one volt greater at this point.

This is illustrated in *Fig. 6* for $R_L = 37{,}500$ and $R_k = 2500$. The operating point is found as before, and falls at $e_c = -10$. Let it next be desired to find where $e_t = -10$ falls. The e_k scale is accordingly displaced 10 volts as shown, and it is seen by matching equal values on the two scales that $e_t = -10$ falls at $e_p = -15.5$. As before, this is a precise determination within the limits of graphical accuracy.

The process may be rapidly repeated for any integral values of e_t which may be of interest for the circuit under analysis.

What do you need...



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If it's the *country's* future, we all need a government and a nation of people with the money to operate efficiently.

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AUDIO ENGINEERING

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Voltage Stabilizers

[*from page 24*]

a-c or d-c, because it is used only to heat a filament, it is equally obtainable from the output of a selenium rectifier stack and the necessary filter, thus offering a high degree of regulation to low-voltage d-c supplies. This type of instrument is also available under the trade name "Nobatron," and an equal degree of voltage regulation is obtainable.

Certain types of thyratron-regulated supplies are noted for the poor waveform, largely due to the steep wave fronts resulting at the firing point of the a-c cycle. These disturbances are usually measurable throughout an entire building in which such supplies are used, and very often seriously affect the performance of other equipment. With the regulation provided by this type of regulator, it is noted that the changes are comparatively more gradual, and the input waveform is not affected. The waveform of these regulators is well within 5 percent harmonic content, and the correction is applied within approximately 6 cycles of the line voltage, or one-tenth second on 60-cps lines.

Where audio circuits are line-voltage operated, considerable improvement can often be obtained by the use of a regulator such as that described here. Inasmuch as the advantages of regulation can be obtained without the sacrifice of a reasonably good input waveform, and at a relatively low cost, it may often be the means of improving distortion and output level characteristics of a sound system. For those users who are limited by an uncontrolled line frequency, this type of regulator offers an inexpensive solution.

Musical Acoustics

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2. Internal noises, often caused by a variable impedance in an electric circuit, or from an induced e.m.f. Noisy carbon-granule and condenser microphones are also guilty.

3. Amplitude distortions from failure of the r.m.s. (or peak) values of response to be proportional to the stimuli under steady-state conditions and at different levels of the intensity of the stimuli.

4. Non-linear distortions which fail in proportionality between instantaneous values of response and stimulus, phase differences being ignored.

5. Frequency distortion from the failure of proportionality between r.m.s. (or peak) values of response to stimulus at different frequencies.

6. Transient distortion. During a transitory period a sound may be subject to any of the foregoing distortions, but it may also have a duration of transitory period which may exceed that of the stimulus, and which may differ in extent for different frequency components. Moreover, transitory oscillations may be produced in the response at frequencies which do not appear as periodic vibrations in the stimulus. The reverberation of rooms may be an influence. Phase differences are usually not acoustically important because, except in extensive filter circuits, the time lag is too short (less than 180°). But even with the most simple mechanical system some transient distortion is, theoretically, inevitable. This is acoustically harmless if non-oscillatory; but if oscillatory at an audible frequency with a long period then objectionable foreign components are heard. The reduction of mass is important to reduce transient distortion, as it is to improve frequency distortion.

7. Other varieties of distortion may come from: playing phonograph records at incorrect or variable speeds; introduced variations in volume level (fading in radio reception, or standing waves in rooms); and transmission or reproduction at an unnatural volume level.

The poor acoustics of many auditoriums give audiences little chance to hear a high quality of live programs. So their hearing gets inured to extraneous noises and reverberations and becomes trained to perform a certain amount of directional discrimination between them. Although they may not be aware of the cause this produces some aural-mental fatigue. The added strain of aural attention might be compared to the visual strain of flicker, although not analogous. The binaural hearing of the listener also adds a certain spaciousness to the live program which is lacking from loudspeakers supplied from a single channel.

This is lost to the single microphone and in no way can it later be restored. Then, too, a microphone has directional limitations. The two ears gather a large percentage of the sounds from reflective surfaces in all directions. Perhaps a multi-directional composite group of microphones spaced according to the listener's ears and located at a favorable listener position, instead of near the sound source, would give an improved result. But it could not equal stereophonic broadcasting and reception.

However, the microphone itself lacks fidelity and uniform response over the desirably broad range up to 15,000 or, possibly 20,000 cycles. It must receive and suffer from high electronic amplification and the balancing of faulty response —electronic tampering.

The very minuteness of the voltages or currents produced by microphones or phonograph pickups make them more difficult to discriminate from the inherent noise due to the thermal agitation of electrical circuits in amplifiers. The threshold of hearing is only about 11 db above the noise by thermal agitation of electrons, so small amplifications make the latter noticeable.

Thermal Agitation

The square of the effective value of such noise voltage components (E) produced by thermal agitation, where (R) is

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the resistance component of the impedance which produces voltages of thermal agitation, and is a function of the frequency (f) which may be expressed as the integral of Rdf between the limits of f′ and f, may be expressed when R is constant by the formula E^2 equals 4 K T R (f′-f) where T is the absolute temperature in degrees Kelvin (equivalent to degrees Centigrade above −273°C.) and K is Boltzmann's constant of 1.374 (10)−23.

The following tabulation of the values for a few conditions will indicate the magnitude of the voltages produced by thermal agitation for a temperature of 80° F. (300° K).

When R is expressed in megohms, instead of ohms, and E is expressed in microvolts, instead of volts, then $E=0.1284\sqrt{VR(f'\text{-}f)}$ where f′-f is the frequency range in cycles per second.

Frequency Range (in cycles)	E in microvolts for Resistance in megohms where R equals		
	½	1	2
5,000	6.42	9.08	12.84
10,000	9.08	12.84	18.16
15,000	11.12	15.73	22.24

The thermal agitation voltage E appearing across the plate load resistance R_0 as a result of thermal agitation in R_0 and temperature T_0 of load resistance and in plate resistance R_p of the tube at the hot cathode (filament) temperature T_f is given by the more complex formula:

$$E^2 = \frac{R_p\,R_o}{(R_p+R_o)^2}\,(T_oR_p + T_fR_o)\;(f'-f)$$

It is evident that both may be factors where many stages of amplification and compensating electronic circuits are involved. But because of subsequent amplification of such extraneous noise the first stage is most important. If the electron emission is not uniform from oxide-coated filaments or is insufficient for an adequate space charge such thermal agitation noises may become troublesome.

Other noise sources from electronic amplifiers are:

1. Hum induced from electromagnetic or electrostatic fields of power line frequencies into filament or heater leads, filter chokes, or amplifier or power transformers; or where there is high impedance to ground (grid circuit of AF amplifier open to grouud or with high resistance to ground).
2. Microphonic noises from the jarring of tube elements by acoustic vibrations from the loudspeaker.
3. Regenerative changes in amplification and frequency response from feed-back coupling in an amplifier.
4. Overloading of amplifiers, causing intermodulation aberrations.

(*To be continued*)

Electronic Organs

[*from page 22*]

The same is true of coupled notes, yet both of these things are accomplished in the electronic instrument.

The attack and decay of the tones of our instrument is very abrupt, as is true in all cases where the output of the tone generator is keyed directly. This has the advantage of permitting great rapidity and accuracy in the execution of musical passages, but it is also a disadvantage, in that it sometimes introduces key-clicks. A characteristic of this type also places a greater demand upon the skill of the performer, since he must play legato passages (consecutively connected notes) with a slight overlapping of consecutive notes, to accomplish a smooth transition from one to the other. However, the effect of a sharp attack diminishes as the reverberation time of the room increases, and is scarcely noticable when the period exceeds one second.

The oscillators used in our electronic organ will maintain their frequencies within plus or minus 0.25% over a period of months, through normal variations in room temperature. A tolerance no greater than plus or minus 0.1% would be highly desirable, but would entail the use of oscillators that were less adaptable to our instrument.

The best of pipe organs is notoriously poor from the standpoint of frequency stability. "Flue pipes" (those whose frequency is dependent upon the length of the air column), which make up the bulk of an organ's stops, are subject to a one per cent increase in frequency for every nine degrees F. rise in temperature. Reed stops, on the other hand, usually exhibit a negative frequency-temperature characteristic. The change per degree is less than for flue stops, and is largely dependent upon the temperature coefficient of the material from which the reed tongue is made.

Changes in Pitch

In addition to frequency drift caused by temperature changes, the stops of a pipe organ will change pitch with changes in atmospheric pressure. Humidity has a pronounced effect upon wooden pipes, due to the swelling of the wood, and its increase in density with the absorption of water. It is possible for these effects to be additive and, when this is true, a pipe may be as much as a half of a chromatic interval out of tune. Most pipe organs are tuned every three to six months, and are often tuned immediately before an event of importance in which the instrument is to be used. Normally, a well-constructed, well-maintained pipe organ will remain within plus or minus 0.5% of the frequencies to which it is tuned, barring severe changes in the weather.

Most of the stops will change pitch by the same amount, so that the relative deviation between stops, and between notes will be slight. The reeds excepted!

The subject of pitch tolerances is a highly controversial one, but is of sufficient importance to warrant the concerted attention of everyone associated with music.

A person having "absolute pitch," or a well developed sense of relative pitch, can distinguish between consecutively played notes differing by as little as 0.2%, under some circumstances.

When a single stop (or a group of stops derived from one rank of tone generators) is played, the stop as a whole may be two or three per cent off of its nominal pitch, without offending the average musically trained listener; this, providing the intervals between the notes of the scale are accurate within about 0.5%.

If, however, two stops differing in pitch by several per cent are played together, the resultant sound will be extremely discordant and unpleasing. The apparent pitch of a note will be midway between the two, and a warbling effect will be produced by the beats between them.

If several more stops are added, and are tuned to frequencies somewhere between the two extremes, the total effect will not be unpleasant, since a grande celeste will have been produced. Generally, however, a grande celeste should not have a band width greater than about two per cent of the nominal frequency.

It may be seen from the foregoing that considerable deviation in pitch can be tolerated when either a single rank or a large number is played in unison, but that the tuning must be critical when two or three stops are used together. This applies, in particular, to the upper registers of the scale, where a slight detuning of a note can cause it to beat with its partner at an objectionable rate. The same percentage of detuning, several octaves lower, would yield a beat too slow to be noticeable during normal playing.

Since an organ must be capable of being played with any combination of stops, and in combination with other instruments, it should be designed to meet the most critical conditions which it is likely to encounter in fulfilling its purpose. This would call for a tolerance within plus or minus 0.1% for each note, with some stops purposely detuned to improve the ensemble. Whether we adhere to this standard, or accept a reasonably satisfactory compromise, is determined, mostly, by the price range within which the instrument has to sell.

We have seen that the functions of the pipe organ can be successfully duplicated in a purely electronic instrument, and that introduction of electronic technique to the art offers means for improvement in performance, which would otherwise be impossible.

The electronic organ described in this issue is but one example of the many types now in existence and under development. Others will be discussed in their turn.

Ultrasonics in Liquids

[*from page 18*]

The construction shown is merely for discussion of the principle. A concentric injection of the air would meet the operating conditions better.

Each such cylindrical slug of water weighs 6 micro-pounds, and the total energy per jet just under 400 watts. Since our present turbojet unit has 80 jets, we could emit 41 h.p. of bullets. This looks definitely promising.

One of the difficulties is the high power density, being about 200 kw/cm^2, so in many cases a splitter or diffuser would be employed to spread the energy.

A great advantage in many cases is the fact that you have no boundary layer problem. A very difficult problem is putting energy into a hundred-pound sand casting. It is difficult to imagine any type diaphragm working into the part of the casting exposed in a pouring gate unless it was inserted while the casting neck was still liquid, and even then the tendency to form gas at the diaphragm would probably reflect all the energy. Shooting liquid bullets at it insures a continuously clean surface, and even if the neck wears down an inch or so in the twenty-minute application, it is thrown away anyhow.

The impedance match problem with bullets is rather unique. If they arrive at a uniform rate they soon set up oscillations in the load due to the elasticity of the load. If they are a third cycle long they give a properly timed push each cycle. Now if they give up their kinetic energy in their own length, as a liquid bullet must when hitting a solid load, the impedance match is automatic, as it is simple matter of them developing a push.

A bullet made of air would give up all its energy, as would one of mercury, if both had the proper velocity, frequency and length. Since the mercury would weigh about 13,000 times as much as the air bullet it would have that much more energy.

There are some points of similarity with a bullet fired from a rifle. Here we have gas coupled to a very dense solid, and from one point of view the coupling is by time. If the barrel is 18 inches long it takes about a millisecond for the bullet to be accelerated to 3000 feet a second. During that time the powder is generating about a thousand kw. In a pistol with a 3-inch barrel, the

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[*Continued from page 45*]

coupling is much less effective as it has less time to work, and we may only reach 500 feet a second.

There are undoubtedly an infinite series of liquid and solid generators waiting to be discovered. It seems to be a good rule to look around for phenomena that are now considered parasitic and troublesome, and analyze their possible application to the generation of ultrasonic power. It is possible almost every such discovery will fill some need in the wide horizons of the ultrasonic world.

Multi-Lingual Systems

[*from page 12*]

of the type shown in *Fig. 4*. It is a 50-ohm ribbon unit similar to those used by interpreters. In its base is a push-button which sets up a light in an annunciator strip before the chairman indicating which delegates wish the floor. Hidden under a lucite bar directly beneath and parallel to the ribbon microphone is a signal light. This light may be flashed slowly by an interpreter whenever an enthusiastic delegate starts racing along too rapidly.

The progress achieved through the use of the facilities described above is commendable if we reflect back to the days before such facilities were available. In earlier times, neighboring nationals participating in trade or treaty conferences scarcely ever left their luxurious hotels. Not familiar with the many tongues spoken, they spared themselves embarrassment and discomfort by assigning deputies. Huddling together after sessions these deputies unscrambled their findings and at long last presented their superiors with fairly complete interpretation. The United Nations delegate of today has no need to participate "by proxy"—he is able to keep himself posted and up-to-the-minute on its proceedings.

Technicana

[*from page 38*[

the internal shields of these types are connected to the cathodes, and are not therefore capable of being grounded, as such shields should be, according to an application note in *Radiotronics*.

Recommended types for cathode-follower serivce are the 6AU6 and the 6BA6, both of which have the internal shields connected to the suppressor grid. Optimum connections for these tubes are with the suppressor grids at the same d-c potential as the cathode, but with an effective by-pass to ground.

SOUND MOTION PICTURE HISTORY

● For those who are interested in the history of sound motion pictures, the series now running in the *Journal of the Society of Motion Picture Engineers* will be found of considerable value. This series, started in the April issue with parts 1 and 2, and continued in the May issue with parts 3 to 7, bringing the reader up to 1930, has been compiled by E. I. Sponable of Twentieth Century-Fox Film Corporation, and will presumably continue in succeeding issues.

Not only does this listing comprise the technical developments with which the industry has grown from the early part-talkies to a full-grown, highly artistic enterprise, but it also includes most of the commercial steps, licensing agreements, and the other data necessary for a complete familiarity with the art, arranged in chronological order. Obviously it is impossible to abstract this information for the benefit of our readers, but those who are interested are referred to the *Journal*.

In addition to this history from the viewpoint of the production of sound pictures, another listing gives the contributions of the theatre service organizations to the development of present day picture quality. This material is also in chronological order, as seen by E. S. Seely, Chief Engineer of Altec Service Corporation, and appears in the May issue of the *Journal*. Between the two articles, a very complete history of both divisions of the industry is presented.

IMPROVING SPEAKER DAMPING

● The following method, long used in disc and light valve recorders, has been found very effective in smoothing out the bass response and improving the damping of a cone loudspeaker.

For example: Connect an 8-ohm resistor (of the proper wattage) across an 8-ohm speaker voice coil, connecting the resultant 4-ohm impedance to a 4-ohm output transformer tap. In this case, 50% of the power is being dissipated in the resistor and the output stage should be capable of supplying this loss and still give the required dynamic power range. Lower values of shunt resistance may be used, making a further improvement in damping, but the resultant parallel impedance must always be matched to the proper output transformer tap to reflect the proper load impedance to the output tubes.

All loudspeakers display a large rise in impedance at the cone and cabinet resonant frequencies and this causes them to continue oscillation after the electrical excitation has stopped. The shunt damping resistor damps this oscillation as it absorbs power generated by the speaker's free oscillation and thus loads it mechanically, arresting the free oscillation.

This system is particularly effective on output stages without inverse feed-back loops, making the output tubes considerably more independent of reflected L. S. impedance variations and improving L. S. damping.—*Charles L. Benson.*

Letters

[*from page 5*]

importance. High-fidelity music is quite probably an acquired taste which will gain acceptance when the public is given an opportunity to hear it more frequently.

In connection with the matter of radio and distraction, an unpublished study made at Wayne University about ten years ago indicated that radio entertainment was generally distracting, but in the case of certain well-conditioned subjects, both quantity and quality of work were increased in the presence of radio entertainment. These tests were made with college students and used simple addition and a content test on fiction reading as the work to be performed. The fidelity of reproduction was comparable to the average home receivers.

It would be interesting to conduct a similar test with variable frequency response to scientifically test the thesis of distraction by high frequencies.

R. B. Nottingham
Sales Manager
Acoustic Products Division,
The Brush Development Comnpany

NEEDLE TALK

Sir:

From time to time we have noticed references in the literature to a phenomenon associated with record reproduction, popularly known as "needle talk." Some of the statements which have been made do not at all agree with some of our own findings, and for what it is worth, we would like to comment on the subject.

Sound radiation from a pickup-record combination can originate either in the pickup mechanism and/or arm, or from the surface of the record itself. The radiation is a function of the projected area of vibration and the amplitude and frequency with which the radiating body vibrates. In pickups with a rigid stylus mounting and high lateral and vertical stiffness, considerable radiation is likely to take place from both the arm and the record surface because of the *lateral* forces involved. This is accompanied by a high rate of record wear and is responsible for the association between record wear and "needle talk." Pickups in this class can be made to have much less sound radiation by decoupling the stylus point from the large masses of the arm and pickup mechanism. The familiar bent shank needle accomplishes this to a large degree. It is obvious, of course, that this method of reducing "needle talk" also reduces high-frequency response.

A good lateral pickup must have enough vertical compliance to be able to respond to the unwanted vertical modulation on lateral records which reaches an appreciable velocity only at very high frequencies. The tiny parts used to support the moving system of reproducers of this type can become fairly efficient sound radiators at frequencies of the order of 5,000 cycles and above. There are two ways of reducing this radiated sound: either to increase the vertical stiffness to reduce the amplitude of motion, or to damp the spring suspension heavily in order to accomplish the same result. Either of these means imposes additional wear on the record, and it appears to be unwise to sacrifice performance in order to quiet the operation.

Since this sound radiation is an indication that the pickup is performing its function of "riding over" vertical modulation, it is not necessarily a sign of undue record or stylus wear. It may be noticed that in several wide-range reproducers the surface noise radiated directly from the pickup head is considerably greater than that which is transmitted through the electrical system. This indicates that the reproducer is not converting the vertical motion into electrical signal output, thereby giving an improved signal-to-noise ratio. At the same time it is responding to lateral vibration of the stylus at frequencies beyond 10,000 cycles per second.

Norman C. Pickering

New Products

[*from page 38*]

FILM RECORDER

Filmgraph Model "HK" permanent recorder and instantaneous reproducer employs 2 reels 16 mm. size 14" in diameter. Capacity up to 5000' of M2 film on which 100 sound tracks are put across the width of the film. Each sound track has continuity up to 3 hours. Selection of recording for playback is facilitated by track and zone (footage) indicators. Machine starts recording automatically the instant the telephone bell (when used with Miles Telemike Inductor) starts ringing or as soon as sound is picked up by the microphone. Designed for long time recording of any voice programs, volume dictation, interviews, piped dictation, telephone recordings, hearings, etc. A single machine can serve an entire organization by remote control.

For further data, write the Miles Reproducer Co., Inc., 812 Broadway, New York 3, New York.

COMPLETE MAGNETIC PICKUPS

Full scale production of Amplifier Corp. of America's recent development of the new and highly popular General Electric DL 1RM 6C Variable Reluctance Magnetic Pickup, complete with tone arm and permanent jewel-tipped needle is now under way.

Now all owners of sound equipment may acquire this cartridge complete in a pickup unit, available in Studio and standard models, for all sizes of records.

These arms offer the user (1) Less record wear because of soft needle suspensions; (2) Less distortion because of the soft spring jewel mounting; (3) Less needle scratch because of response only to lateral vibrations; (4) Less needle talk because of the very light and small stylus assembly.

Additional features in the Studio Transcription Model (160GE) include a new type of swivel base for raising or lowering to desired height; low frictional pivots for constant needle pressure; low vertical inertia for ideal tracking, even on poor records; easily maintained spring adjustable needle pressure of 1 oz.

For literature address inquiries to Amplifier Corp. of America, 398 Broadway, New York 13, N. Y.

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