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Vol. 31, No. 5

COVER ILLUSTRATION

Operator at sound console in display room at the Camden plant of the Radio Corporation of America. This sound system is used to bring music to workers on the job, to broadcast special announcements, and to page employees.

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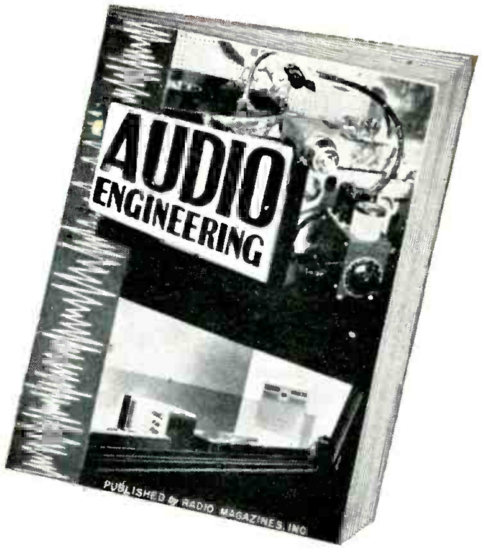

Each issue covers 5 or more of these subjects

1—Broadcasting

2—Transmitter and receiver manufacturing

3—Sound-on-film equipment

4—Recording (disc, wire and tape)

5—Public address

6—Industrial sound equipment and applications

7—Acoustic treatment of studios, rooms, auditoriums, etc.

Our editorial policy has aroused tremendous interest and acclaim among engineers prominent in the industry. We are proud to announce that co-operating with the editors of AUDIO ENGINEERING are such outstanding authorities as:

HOWARD A. CHINN
Chief audio engineer for the Columbia Broadcasting System

J. P. MAXFIELD
Authority on sound engineering, Bell Telephone Labs.

JOHN P. COLVIN
Audio Facilities Engineer, American Broadcasting Company

GEORGE M. NIXON
Ass't Director of Technical Development
National Broadcasting Company

who are members of the editorial advisory board.

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Letters

From a Well-Known Engineer

Sir:

I was greatly impressed by your first issue of Audio Engineering. When I first heard of your decision to convert "Radio" to "Audio Engineering" I was inclined to feel that you were converting to a rather specialized field. However, judging by your first issue, you will be able to keep the interest of general engineers as well as the audio specialists.

Howard A. Chinn's article on "Magnetic Tape Recorders in Broadcasting" was very interesting to me. Also S. Y. White's method of "Ringing a Bell at its Fundamental Mode". In connection with the article "Too Much Audio", you were probably interested to read in today's paper about the deafening of eleven firemen by some extremely loud sound connected with fire-fighting in the Brooklyn tunnel. . .

Murray G. Crosby,
Paul Godley Co.,
P. O. Box J,
Upper Montclair, N. J.

Sir:

. . . The simple fact is that the men who read *Audio Engineering* are for the most part imaginative characters bubbling over with ideas, theories, experiments, questions, etc. Many of them, I am sure, could write excellent articles for Audio Engineering; they should be encouraged and solicited by the editors. Most of them will have *ideas for articles* which they cannot or do not wish to write themselves, but which the editors could assign to others for investigation, experimentation (if necessary), and subsequent reporting; such ideas should also be solicited. For example, I should like to see a detailed article describing the Miller system of sound recording on film.

This brings up another point which is a question of editorial policy, namely: should the inventor or manufacturer of any given device be the one to write the article describing it? Personally, I would prefer to see it written by a competent technical man who has no axes to grind and who has the laboratory facilities to test the subject matter . . . The whole point is, of course, to get as accurate and objective an appraisal of new developments as is possible. . .

To achieve the greatest interest, the editors might well emulate the technique of *Look* magazine, in which the person being interviewed or written about is given the opportunity to answer the article in the same issue, point by point if he wishes. . .

Gerald Shirley
474 W. 238th St.,
New York City.

Many thanks for your suggestion which will be carefully studied. We need good articles on any subject within our field—new ideas, methods, simpler ways of doing difficult things.—*Ed.*

Deep in the Heart of Texas

Sir:

. . . Audio frequency engineering is of interest principally to broadcast studio and recording engineers; certainly, very few others. Personally, I think our modern broadcasting tendencies are a national disgrace . . . what with singing commercials and vulgar references to "body odor" . . . Consequently, a magazine which even hints of broadcast studio technique is very distasteful to me . . .

R. E. Lovejoy,
Route 3, Box 290D
Mesquite, Texas.

For ex-reader Lovejoy, Mum's certainly not the word!—*Ed.*

Sir:

I wonder if you could do something about the situation concerning phonograph record grooves and stylus tips. Perhaps you might try a slow program of "propaganda" in an attempt to correct a condidition whereby a half-dozen or more groove shapes exist and record men use abrasives in manufacturing while stylus manufacturers keep making harder tips to fight the abrasion. . .

Ted Powell,
Maspeth, L. I.,
New York.

We're all for it. Anyone have any specific ideas on standardization of groove shapes? We're glad to open this question to general discussion.—*Ed.*

EDITOR'S REPORT

IT'S JUST A MONTH since we put to bed our first issue of AUDIO ENGINEERING, and we thought you might be wondering how we are making out. The first few hundred copies were rushed to the Chicago Radio Parts Show, where we had a booth, and we expected such a quantity would take care of requirements for the entire period of the show. The interest was so great, however, that this supply was practically exhausted in a few hours.

Two days later, on May 15th, *Time* magazine hit the newsstands with its entire science section devoted to a discussion of S. Young White's article, "Too Much Audio", from our first issue. The following week John Patric phoned in, telling us he was doing a story for the *Saturday Evening Post* on the Wagner-Nichols recorder article. Next, a cable from *Electronic Engineering* in London, requesting permission to reprint our lead article by Howard Chinn.

The New York Public Library, our office, and our representatives have been swamped with calls for copies and further information on the magazine. Our circulation department has been enlarged to take care of the flood of subscriptions and inquiries. Celebrities—even Dr. Lee de Forest—have written in, expressing keen interest in AUDIO ENGINEERING. Some letters from others in the field are reprinted in this issue.

The variety of occupations represented by those who have written to us is really astonishing. While the majority are from those in the radio and sound recording fields, we have also heard from Oak Ridge, Fisher Bodies, a sprinkler manufacturer, several universities, foreign governmental departments, and even the secretary of the Long Island Duck Farmer's Association. We believe that never before has an issue of any technical magazine in this field received such widespread attention.

All this, of course, is nice to know, and speaks well for the work of the editorial advisory board and our staff. But we realize that there is still considerable room for improvement. We shall add articles of special interest to the public address specialist, and for the radio receiver engineer working on the audio sections of sets. One reader suggests the 6AS7 as an interesting possibility for a low-distortion output stage. It's worth investigating.

COMING ARTICLES

J. P. Maxfield's long-awaited article on microphone placement is now complete and is being cleared through Bell Labs. Howard Chinn has sent in a fine article on the use of single plugs for broadcast apparatus. His articles are always a delight to an editor because they are so carefully and neatly prepared. Each sentence is polished until it gleams, and there is practically nothing to do but mark for type size and send to the printer. George Nixon is down with the mumps, but will be back at work shortly. Jack Colvin has just brought in the first of a series of articles on planning broadcast studio installations. Jack has done a great many of the more elaborate jobs for the American Broadcasting Company, and the methods he has worked out makes complex layouts seem simple. Art Schneider is finishing his article on the sound installation for the United Nations headquarters in Flushing. S. Young White will be in again with another of his unusually fascinating articles which have aroused so much discussion. Incidentally, he's the White of the Loftin-White combination, holders of 175 patents in the radio field, including that of the universally used padding condenser. Commander Loftin is no longer active in the field. Norman Pickering will reveal some of the excellent engineering behind the design of his pickup. W. W. Dean and L. M. Leeds of the General Electric Company will present technical details on the new G.E. limiting amplifier which they developed. And there are many others which cannot be announced at the moment.

Many readers have written in, suggesting specific subjects for articles. In some instances, these have already been well covered in the literature. When such is the case, we are only too glad to inform the writer. If it's something new, and you know the subject well, write us a brief synopsis and perhaps we can make a definite assignment for a full-length article for which, by the way, we pay top rates.

LISTENER PREFERENCE TESTS

In this issue listener preference tests conducted by Moir in England and a report on Dr. Harry F. Olson's tests in this country are presented. Each obtains different results, but it is significant that Moir used reproduced sound while Dr. Olson employed "live" music in his tests. More tests are needed, conducted under carefully controlled conditions, before any accurate conclusions may be reached. Another article on this vitally important subject is now in preparation. The article on the high-quality amplifier represents a production model of that originally demonstrated by Lincoln Walsh before the Radio Club of America some years ago. At that time development of associated equipment, such as pickups, tuners, and loudspeakers had not approached the standard reached by amplifying apparatus, so there appeared to be little point in carrying amplifier design to such a high standard of performance. With the greatly improved equipment now available, such amplifiers acquire fresh interest as a link in a complete, high-quality, reproducing system.

—*J.H.P.*

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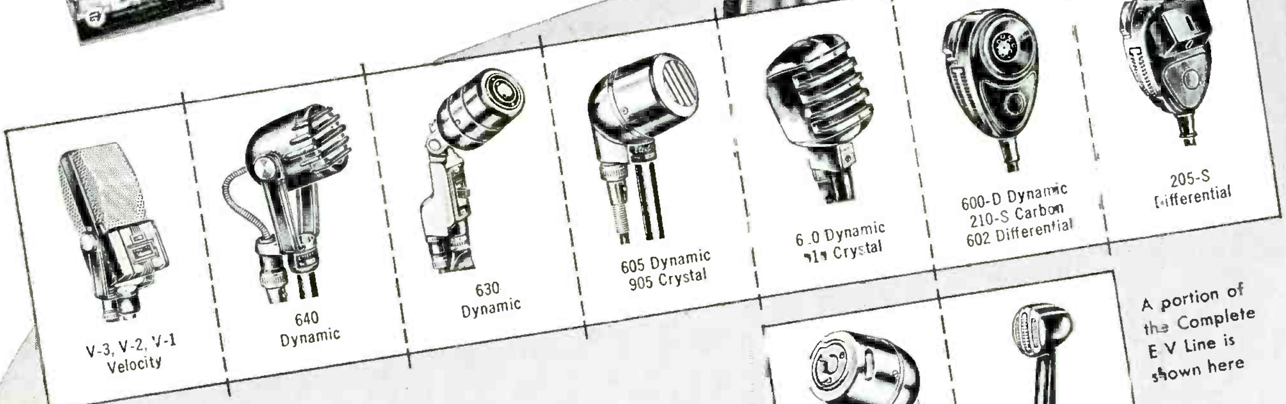

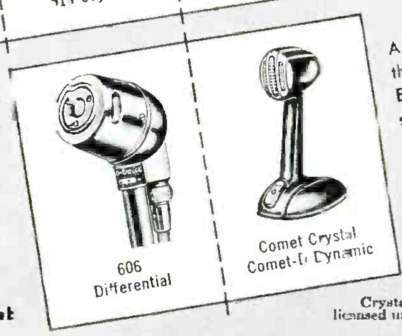

A portion of the Complete E-V Line is shown here

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View of the new, high-quality amplifier described in this article.

High-Quality Audio Amplifier With Automatic Bias Control

Exceptionally low distortion and uniform response over a wide frequency range are features of this audio amplifier. Power triodes operating with automatic bias control are used in the output stage.

J. R. EDINGER
Brook Electronics, Inc.

The simplest definition of a high-quality amplifier is "An amplifier which reproduces sound indistinguishable from the original." Broken down into its technical elements, it requires that:

1. All audible frequencies shall be amplified uniformly.
2. No new audio components shall be introduced. Harmonic and intermodulation distortion should be at such an extremely low value that they are not detectable in themselves and, more important, do not contribute to "listening fatigue."
3. The gain shall be uniform at all signal levels. Amplitude distortion shall be inaudible.
4. Transient waveforms shall be transmitted without distortion.

If one or more of the above elements is lacking, the sound is not natural and cannot be called high quality. For instance, a flat audio characteristic extending from 20 to 20,000 cycles, if it is not free from distortion, is actually disagreeable and far less acceptable than a much narrower audio range, from 100 to 4000 cycles, with the same amount of distortion. When distortion, or noise such as static, hiss or needle scratch, is present in any part of the system the most pleasing sound is to be obtained with some attenuation of the high frequencies.

To produce a high-quality sound system, distortion of all kinds must first be reduced to the vanishing point, and then the audio-frequency range must be widened to the full limits of audibility. Then, and only then, will the reproduced sound be comparable to the original.

Listening fatigue is a very important, but intangible, factor in high-quality reproduction. There is frequently a strong desire to stop listening, even though a good reproducing system is being used. This is evidently a result of having excessive intermodulation and high-order harmonic distortion, which is not readily detectable during listening. It is, therefore, very necessary in a high-quality amplifier to keep distortion to as low a value as possible, even though it may not be apparent.

Low-mu triode tubes were chosen for the Brook high-quality amplifier, be-

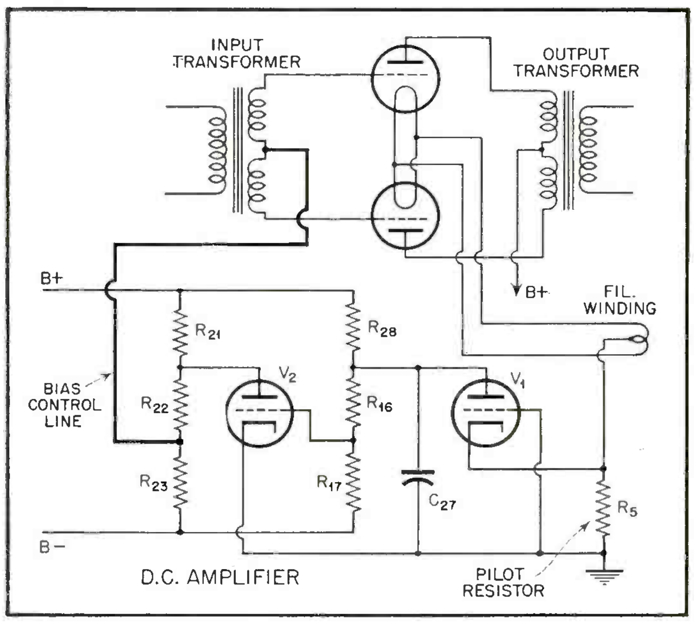


Fig. 1. Fundamental circuit for automatic bias control. See text.

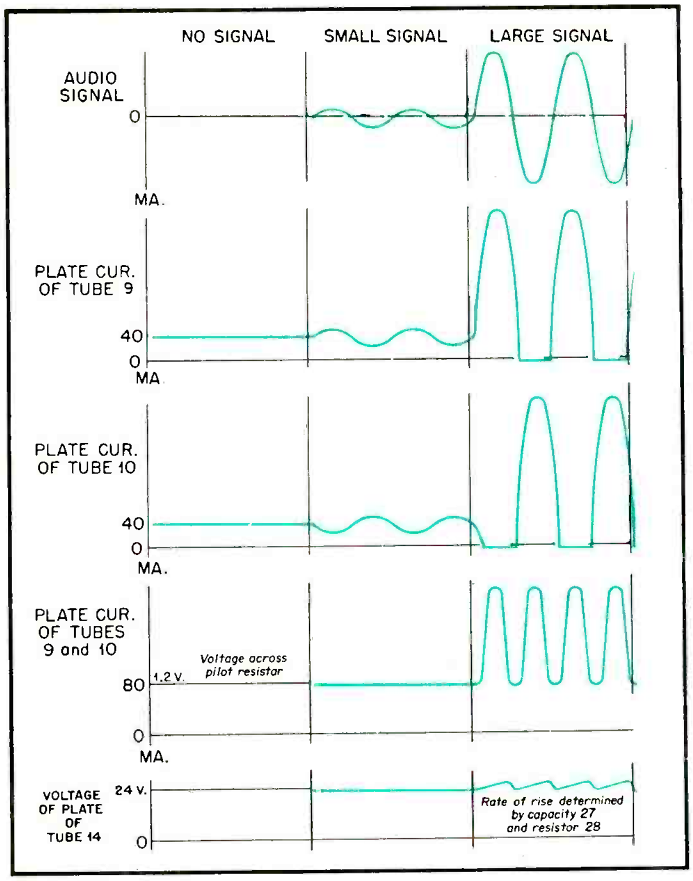


Fig. 2. Manner in which the plate currents of the output tubes vary for small and large input signal voltages.

cause triodes inherently are almost completely free of high-order distortion. Pentodes and beam power tubes have slight kinks in their characteristics curves which produce exceedingly small amounts of very high order harmonic and intermodulation distortion. These amounts, while hardly measurable, are yet believed to be responsible for the listening fatigue.

A number of tests have shown that when a good speaker system is used, extending the frequency range from 8,000 to 14,000 cycles produces a very desirable increase in listening pleasure when triodes are used.

The same tests were made with the best amplifiers available using beam power tubes and pentodes employing feedback. The increased range was definitely irritating. There was the desire to stop listening. The only evidence found to produce this listening fatigue was intermodulation and high-order harmonic distortion.

The same tests were again repeated with a triode amplifier, but the triode amplifier's bias was purposely set to give a small step or kink in the tube's characteristic curve. This amplifier then also produced what we call listening fatigue.

Automatic Bias Control

The problem was to build an amplifier where the bias would always be of correct value. The Brook amplifier accomplishes this by using automatic bias control.

The automatic-bias-control (a.b.c.) circuit used in this amplifier is a recent development. It was developed and patented by Lincoln Walsh for use in audio amplifiers and other electronic devices shortly before the war.

The purpose of a.b.c. is to operate the amplifier at the desired condition for maximum distortion and highest efficiency. The a.b.c. circuit reduces the harmonic distortion and the intermodulation distortion values to a new low.

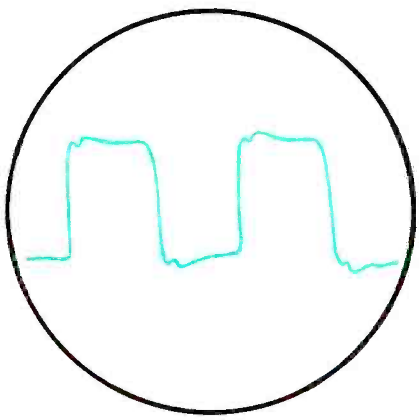


Fig. 3. Oscillogram of 5000-cycle square wave after passing through this amplifier. This indicates good transient response.

This amplifier with a.b.c. operates as a pure class A system up to 5 watts output; then there is a smooth transition to class AB_2 operation up to 30 watts output with a pair of 2A3 tubes in push-pull, or 6B4G equivalents.

The proper choice of elements in the a.b.c. circuit results in maximum power output, minimum distortion, and highest efficiency. Under static or no-load conditions, the plate current of the two push-pull tubes is maintained at the desired value for good tube life. Under dynamic conditions or load up to maximum undistorted output, the plate current will rise substantially, but the grid voltage remains at its no-load value. As shown in *Fig. 1*, this is accomplished by transmitting the output tubes' plate current through this resistor as a control by passing it through a d-c amplifier, and taking the output of this amplifier to automatically control the voltage applied to the grid-bias circuit of the push-pull output tubes.

The pilot resistor, R_5, is connected to the midpoint of the filament winding so that the plate current of the two output tubes flows through it. The voltage accross this pilot resistor is impressed between the cathode and the d-c amplifier tube.

The resistor network through which the plate current of V_1 is fed is of high resistance, so the plate voltage is approximately the grid voltage multiplied by the mu of the tube. The grid voltage of V_2 is intermediate between the variable positive voltage of the plate of the first tube, and the fixed negative voltage derived from the power supply.

In this manner, the grid of V_2 is maintained at a negative voltage, which is desirable, and yet directly coupled to the plate of V_1 which is at a positive voltage.

The tube V_2 and the resistor network R_{21} R_{22}, and R_{23}, form a similar d-c amplifier stage. The output is conducted by the a.b.c. control line to the center tap of the secondary of the interstage transformer, substantially unchanged by the cathode driver to the output tubes' grids, thus controlling the output tube plate currents.

In operation, as the bias on the grids of the push-pull output tubes is made more negative by the a.b.c., the plate current of these tubes is reduced to a value at which the system finds equilibrium.

When a signal is impressed on the amplifier the plate currents vary as shown in *Fig. 2*. For small signals, the change in the total average plate current is negligible. For large signals, the total current increases twice per signal cycle, and for very large signals, each tube alternately goes completely to cutoff, and the current flows through only one tube during part of a signal cycle. But for short periods during the signal cycle when the signal voltage is crossing the zero axis, the plate current of each tube is very close to its zero signal value, as is the sum of the plate currents.

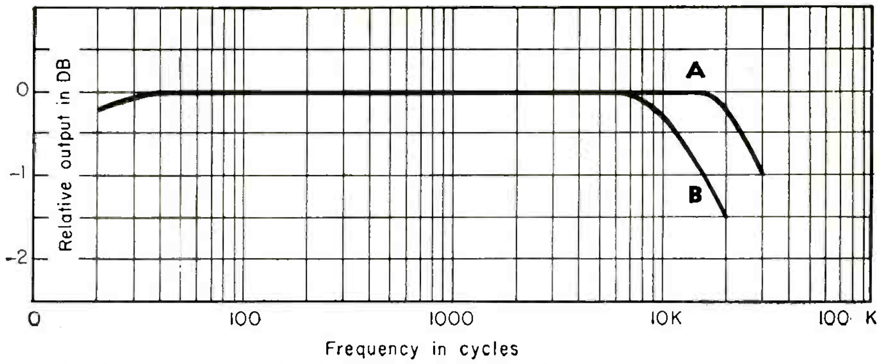
For best operation, it is desirable that the instantaneous minimum of the sum of the plate currents be held substantially constant at the zero signal value. This a-b-c circuit is therefore designed to utilize this instantaneous minimum of plate current as the control factor in setting the grid bias, and is substantially unaffected by changes in the average current due to increases in the plate current which occur during the signal cycle.

When a small signal is impressed, the average current through the tubes, and therefore, through the pilot resistor, does not change. (Class A operation.)

When a large signal is impressed, the total plate current rises, as shown in the fourth curve of *Fig. 2*, increasing the negative voltage on the grid of V_1, in respect to its cathode, and the plate of V_2 voltage tends to rise. But you will note there is a condenser, C_{27}, which is shunted from the plate of V_1 to ground. This condenser and R_{28}, which is of a very high value, control the time constant of this circuit. So as the grid of V_1 becomes more negative, the voltage of its plates cannot change instantaneously but rises slowly at a rate determined by C_{27} and R_{28}.

Until the plate voltage of V_1 reaches a value approximately μ times the grid-cathode bias, no plate current flows, and the current flowing through R_{28} flows into the condenser, increasing its voltage. The time constant of C_{27} and R_{28} is quite long relative to any signal cycle, and so before the voltage builds up appreciably, the signal approaches the zero axis, the plate current of the output tube drops to its zero signal value, and the grid bias on V_1 drops to its zero value. The plate of V_1 again conducts and discharges C_{27}, down to its original voltage corresponding to zero signal as shown in the last curve of *Fig. 2*. As the charge on condenser C_{27} has too short an interval to build up its voltage appreciably, its average voltage remains substantially at the value for zero signal and does not change appreciably when the signal increases from zero to a very large value. The plate voltage of V_1 impressed on the voltage divider formed by resistors R_{16} and R_{17} results in a voltage being impressed on the grid of V_2 as stated before, which varies proportionately to the plate voltage of V_1, but negative relative thereto, so the grid of V_2 is negative.

The result is the plate voltage of V_2 varies with that of V_1, but in opposite sense, and in larger amounts due to the amplification of V_2. Similarly, the plate voltage of V_2, impressed on the voltage divider R_{22} and R_{23}, results in a voltage known as the bias control to the amplifier output tubes. This a-b-c voltage varies proportionately with the voltage on the plate V_1, but negative. The a.b.c. is directly controlled by the instantaneous minimum of total plate current of the output tubes, which is the desired condition for maximum power output and minimum distortion, the highest efficiency.

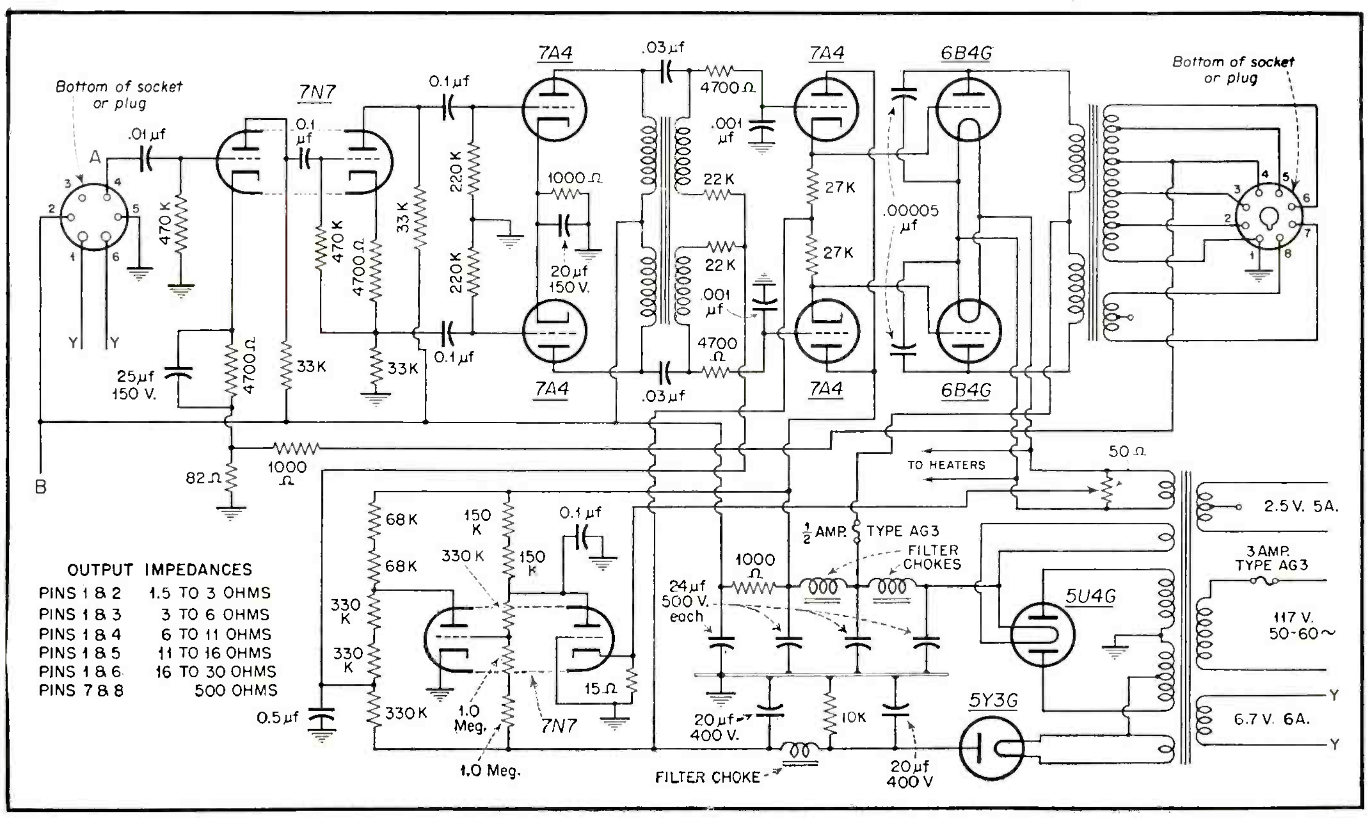


Fidelity response curve of high-quality amplifier. In (A), output power is one watt. Curve (B) shows response when the power output is 30 watts.

Transients

Transient response cannot be neglected in a high-quality system. [*Continued on page* 41]

Fig. 4. Complete schematic of the Brook high-quality amplifier. Special interstage transformers are required.



Adapting Paper Tape Recorders For Broadcasting

RICHARD S. O'BRIEN*

Describing the modifications required for greater utility in broadcasting service.

THE INHERENT ADVANTAGES of magnetic recording on coated paper tape for broadcasting and other uses have been described previously.[1,2] It seems likely that this new medium will be used for many spot recording applications. At present the technique is sufficiently advanced to be readily applied in many of these cases. In fact, the need for a recording process of this type has led to the immediate utilization by broadcasters and others of the first recording equipment commercially available in this country.

This recorder, the Brush BK-401, was designed for home-recording use and as a result some of the refinements usually found in professional equipment were not included. A number of simple modifications have been evolved, however, to partly make up this lack without altering the basic recorder design. These modifications, together with several related maintenance and operating factors are discussed in this article. In addition, an automatic-control modification is described whereby the recorder and a radio receiver can be arranged to enable delayed-listening—a feature often desired for home or office installations.

The modified recorder will prove quite serviceable for interim use pending availability of professional-grade equipment. Its use will enable operating

*Columbia Broadcasting System, Inc.

[1] "Magnetic Tape Recorders in Broadcasting", H. A. Chinn, AUDIO ENGINEERING, May, 1947.

[2] "Recent Developments in the Field of Magnetic Recording", S. J. Begun, *Journal of Soc. Motion Picture Engineers*, January, 1947.

Fig. 1 (left). Brush BK-401 magnetic paper-tape recorder adapted for broadcast recording applications. The standard illuminated type VI is mounted at a slight angle to give good visibility with the VI selector-switch and earphone monitor jack mounted below it. The VI amplifier chassis is located behind the VI meter inside the cabinet. In the unit shown, the VI is used only for recording as only a low-level playback output is provided. However, where the output circuit is arranged to feed a transmission line directly, a fourth switch position may be provided to enable VI and earphone monitoring of "line" output. Fig. 4 (right). Brush BK-401 recorder adapted for automatic operation in home or office installations. The recorder and a radio receiver associated with it may be operated normally with the auto-manual switch set to "manual" as shown. With this switch set to "auto", they may be automatically operated to record any two 15-minute periods as preset on the built-in clock/timer. The playback/radio switch enables either the recorder playback output or the normal radio signal to be fed to the receiver audio amplifier and loudspeaker.

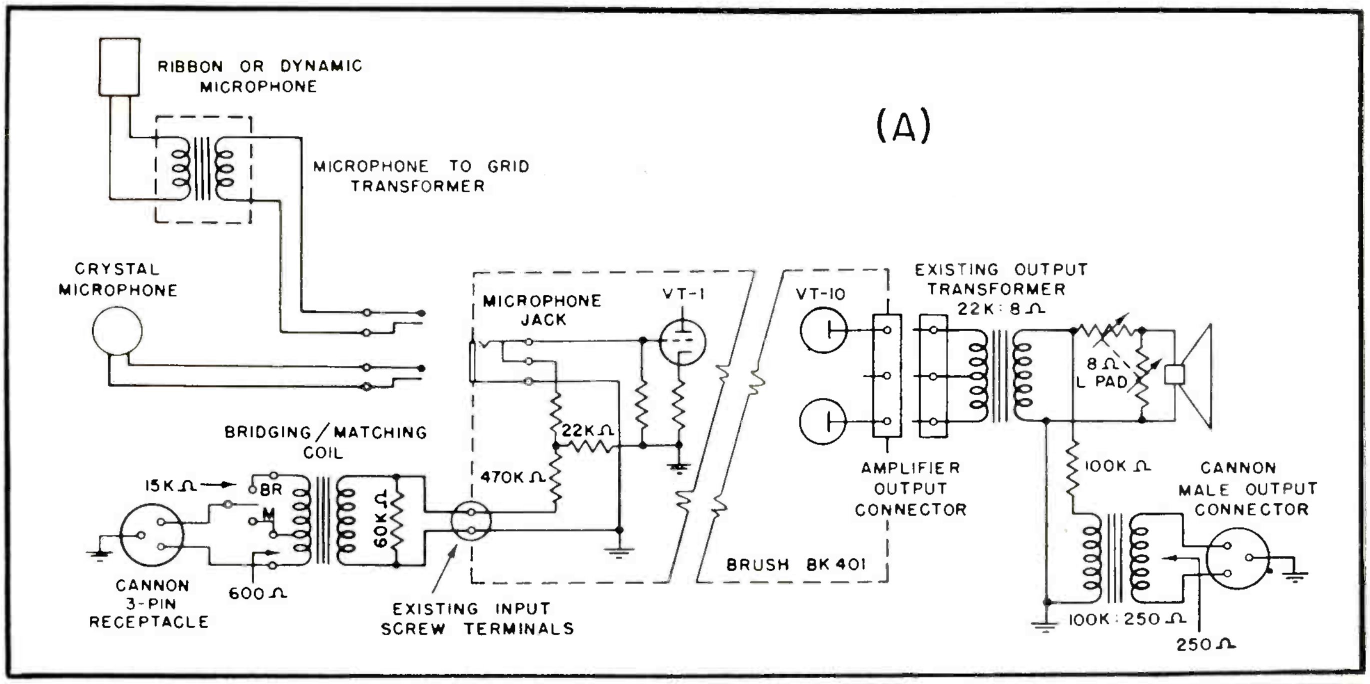


Fig. 2A. Input and output coupling circuits for adaptation of Brush BK-401 recorder to broadcast use. A tapped-primary input transformer is inserted in the existing bridging input circuit with a switch to provide either matching or bridging impedance at the input receptacle. The microphone input channel may be used with the crystal microphone or with a low-impedance microphone by feeding into the jack through a cable-type microphone-to-grid transformer. A low-level output, which can be coupled directly to an audio mixer position is provided by bridging across the speaker voice-coil circuit. The 8-ohm L-pad enables adjustment of recorder speaker volume without altering the output signal level.

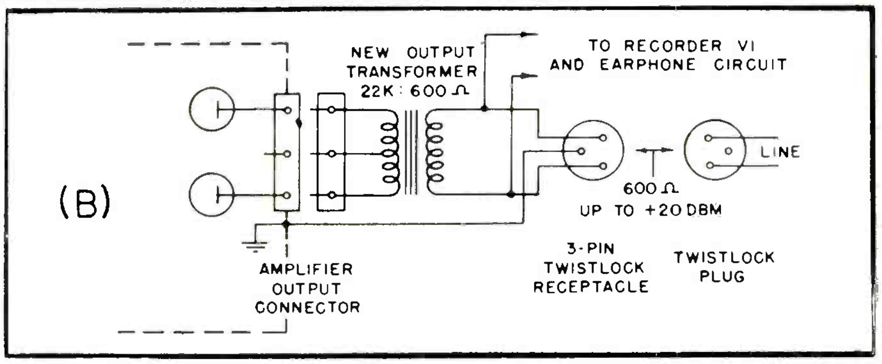


Fig. 2B. Alternative high-level output coupling circuit for Brush BK-401 recorder. A new output transformer is plugged into the amplifier output in place of the originol speaker coupling transformer. The secondary is coupled to a 600-ohm line through a twistlock connector. The VI and earphone monitoring facilities built into the recorder for recording may be used for monitoring this high-level output.

personnel to become familiar with the medium. For example, certain problems of editing, splicing and general handling of the tape exist regardless of the type of recording machine. In general, the fidelity of the present recorder will be found surprisingly good.

Modifications For Broadcast Applications

Magnetic tape-recording is an ideal medium for special events pickups, delayed broadcasts, interviews, etc. Many such jobs can be handled readily with the present recorder. Of course, the unit must be equipped with input and output coupling circuits which allow it to be fitted into the broadcast station audio system. Also, replacement of the electron-ray volume indicator "eye" with a standard VI, and minor circuit refinements to reduce noise level and susceptibility to electrical interference are desirable. These "broadcast" modifications have been applied to the recorder shown in *Fig. 1.*

Various input and output circuits can be used depending on the intended use of the recorder. The gain of the built-in amplifier system is sufficient to handle input signals of microphone level (−50 dbm) or higher, while output levels up to +20 dbm may be obtained. Both input and output circuits, except in the case of the existing high-impedance microphone input, should provide isolation, should supply the proper impedance, and should be workable with either balanced or unbalanced external circuits. In general, this calls for transformer coupling.

A typical coupling arrangement is shown in *Fig. 2A.* The input transformer primary is tapped to provide either bridging or matching impedances. With this input system, the recorder can be bridged across 600-ohm balanced or unbalanced circuits and will be fully modulated by signal levels from −29 to +10 dbm. A signal level as low as −43 dbm may be fed into the matched impedance coupling. The existing crystal microphone input is retained, and it should be noted that a microphone-to-grid cable-type transformer can be plugged into the microphone jack to accommodate velocity or dynamic microphones.

In the output circuit, a transformer having an impedance ratio of 100,000/250 ohms, in series with a 100,000 resistor, is bridged across the output transformer voice-coil winding. The arrangement provides an output which appears to have a source impedance of 250 ohms and results in an over-all loss of 56 db. Thus, with an audio level of +10 dbm in the speaker circuit, the coupling circuit output would be −46 dbm—a level which can be applied directly to a microphone-input position of an audio system. The recorder speaker volume can be independently attenuated by means of the L-pad control, once the recorder output level has been set. Other circuits may be used to provide low, medium or high-level outputs from the recorder. In one alternate output coupling, shown in *Fig. 2B,* the existing output transformer and speaker are replaced by a new plate-to-line transformer plugged into the amplifier output receptacle. With this circuit, levels up to +20 dbm can be fed directly to a transmission line. Lack of a loudspeaker is offset by provision for earphone and VI monitoring of the line level output.

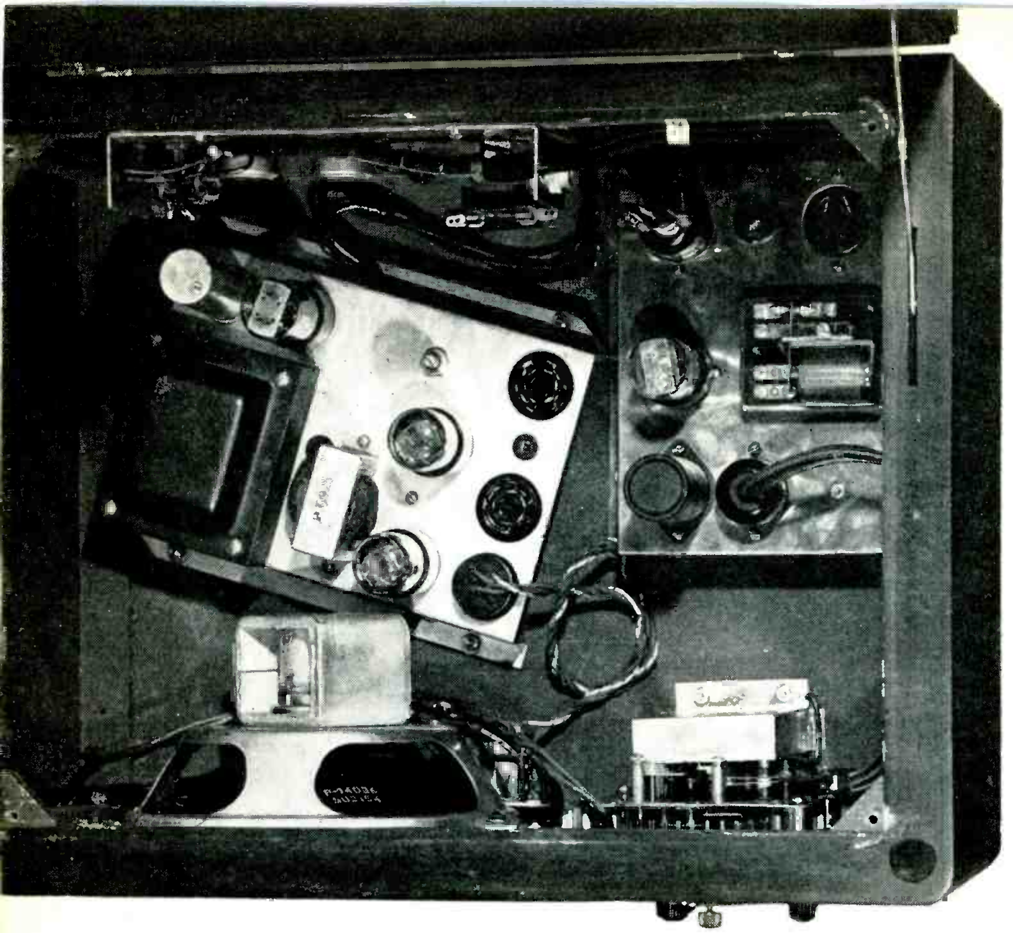


Fig. 6. Inside view of Brush BK-401 recorder adapted for automatic operation, showing the clock/timer with its motor shield in front at the right; the automatic control and starter unit at the right-rear; the recorder power chassis orientation required to minimize hum pickup, and the audio connection panel along the rear of the cabinet. The unconnected fuse visible at the rear is a spare, the clip having been removed from its original mounting inside the power chassis. The system fuse is mounted on the control chassis. The receiver power connector is plugged into the standard a-c receptacle at the rear of the control chassis.

It should be noted that additional precuations may be required in installations where several recorders must be operated in close proximity. Thus, the power chassis orientation in a particular recorder must be arranged to produce minimum hum pickup in both that recorder and its neighbors. Also, motor-circuit switch spark suppression is recommended so that adjacent machines can be run independently without danger of introducing pops in the "on-air" circuit.

These various modifications can be installed in the existing recorder cabinet. *Fig. 1* shows the placement of VI meter and switch and an earphone monitoring jack. The input circuit and connector can be mounted on a narrow vertical panel on the left side of the rear of the cabinet, as viewed from the rear. The output connector and a speaker volume control can be mounted on a similar panel on the right side. In addition, the power input lead can be brought in through this panel to an extractor-post fuse mount and thence to the recorder power chassis. This makes the fuse accessible and also clears the path for enclosure of the space between these two panels with a sheet of perforated-metal grille to protect the innards of the recorder. This grille can also be used to mount a ventilating fan which is recommended where the recorder is to be used for periods exceeding one hour. An induction fan motor should be used to avoid commutator hash, and a type of motor in which the coils are completely enclosed by the field structure is essential to avoid hum fields. A four-pole, 1600 rpm motor turning a 5" four-bladed fan will easily hold the recorder temperature rise to 30°F above ambient. The fan should be located as near the main recorder amplifier strip as possible.

A recorder which has been electrically and mechanically modified as discussed will be serviceable in many applications. However, a more thorough repackaging job should be considered where a more conveniently shaped and more rugged unit is needed for field use.

A standard VI may be coupled into the recorder as shown in *Fig. 3*. An audio signal is taken from the unbypassed cathode of the final stage in the "record" amplifier, to obtain an output signal isolated from the 30 kc r-f bias applied in the plate circuit of this stage and, incidentally, to make use of the low impedance transmission circuit thus provided. The signal is amplified and equalized to compensate for the pre-recording low and high-frequency boost introduced in the recorder and transformer-coupled to a 600-ohm termination. Either a standard VI panel or the built-in meter circuit may be plugged into the amplifier output. "Test" level at 1 kc requires an input to the VI amplifier of 0.2 volts, rms, and corresponds at the VI meter circuit input to a level of +10 dbm. "Program" level corresponds to a +4 vu* level at this point. The amplifier may be assembled on a small chassis and fitted into the recorder cabinet. Power may be applied to it from the vacated 6E5 socket, and if desired the audio input connection may be made via an octal adapter inserted under the final "record" amplifier, 6SJ7, tube. Use of the VI enables more accurate control of recording level and leads to improved quality and uniformity of recordings.

The inherent noise level of the tape recording process is low. However, because of the extremely low output from the pick-up coil and the proximity of copious sources of hum, several precautions are necessary to obtain full advantage of this feature. In the Brush BK-401, the power chassis is usually oriented so the null in the power transformer field is in alignment with the pick-up coil. This orientation should be checked with the recorder running to make sure it is optimum. A light-gauge magnetic shield placed around the front and exposed side of the take-up motor is often effective in further reducing the noise level. An adjustable heater circuit ground point (100-ohm potentiometer by-passed to ground) will provide a range of adjustment to buck out certain hum components. Separation of all low-level amplifier grounds from other ground returns and routing of this audio ground circuit independently to the first amplifier tube ground point is of further benefit. In conjunction with this change, these same audio circuits should be rerun in shielded leads to reduce susceptibility to pick-up from strong, nearby r-f sources such as diathermy and vhf transmission.

*The term vu applies to standard VI meter indications on program material whereas dbm refers to indications on steady test tone signals. The 6 db difference in "test" and "program" VI settings allows only a 6 db headroom for program peak excursions, this reduction from the usual 10 db resulting from a compromise of noise level and distortion considerations.

Modification For Automatic Operation

The modified Brush BK-401 recorder shown in *Fig. 4* may be operated automatically, controlled by the built-in Telechron radio timer. The recorder input may be supplied from any suitable audio source but the system shown is arranged especially to work with a radio receiver.

For automatic operation the timer can be preset within an 11¾ hour period preceding the intended recording time. Within this limit, any two fifteen-minute periods, either consecutive or separated, may be recorded; the thirty-minute total recording time being set by the capacity of the tape reels. Presetting of the system requires the user to go through all normal operating steps preliminary to switching over for automatic operation. A single switch takes care of all circuit changes in switching from *Manual* to *Auto,* the only remaining step being the setting of the timer.

The modifications described above concerned with reduction of noise level and interference pick-up should be applied to the automatic home-recorder adaptation. In addition, a special control circuit is required to automatically start the recorder. Also, a suitable means of coupling the recorder and radio receiver audio systems together is required.

In the control circuit, shown in *Fig. 5,* power is supplied to the timer motor continuously. With the auto-manual switch set to *Manual,* power is supplied continuously to the recorder and radio which may then be turned on and off independently by means of their normal power switches. With the switch set to *Auto,* power is supplied to the radio and recorder through the timer contacts only during preset "on" periods. At the start of an "on" period the starting circuit supplies a one-second pulse to the recorder, following a 20-second delay which allows the amplifier tubes to warm up. The short starting pulse is sufficient to put the tape in motion, pulling in the pressure-operated forward-stopping switch to maintain the motor power circuit. The action is identical to that obtained by momentarily pressing the recorder "start" button and then releasing it so the interlock feature of the held-in stopping switch is not lost.

The relay circuit is one of many possible for the cycle described. As the 117Z6 rectifier reaches operating temperature, relay *1* closes, applying power to the recorder drive and take-up motors. At the same time, relay *1* opens a short-circuit jumper across relay *2,* allowing relay *2* to close, permanently opening this short circuit so as to hold its circuit closed. In closing, relay *2* de-energizes the 117Z6 circuit, relay *1* holding in for about one second while the charge in condenser C is dissipated. When the power is removed at the end of a recording period, relay *2* is de-energized and the circuit returned to starting condition.

The layout of the control chassis and its placement in the recorder cabinet are shown in *Fig. 6.* The timer motor is covered by a light gauge magnetic shield which prevents the timer motor shield from being picked up by the playback head.

In *Fig. 6,* the strip along the rear edge of the recorder cabinet is the connection

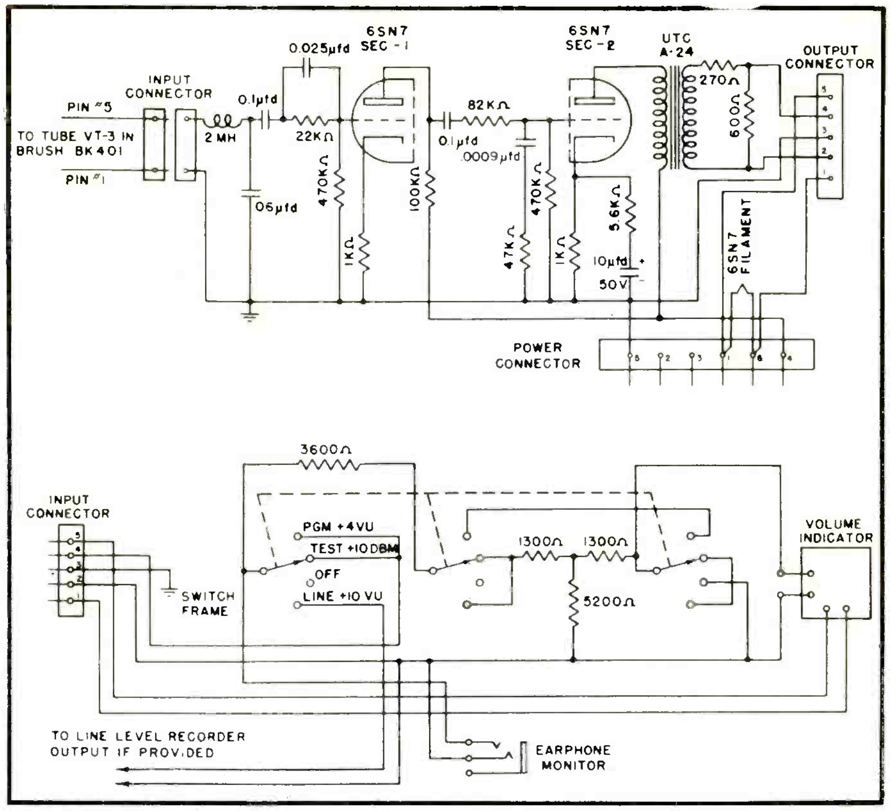
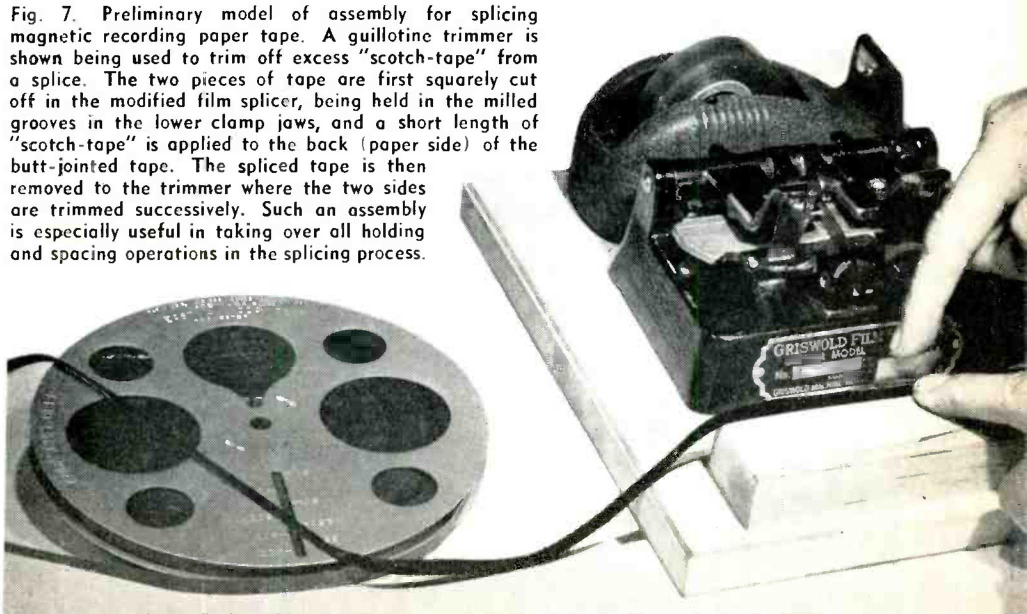


Fig. 6. Inside view of Brush BK-401 recorder adapted for automatic operation, showing the The two-stage amplifier-equalizer circuit is assembled on a small chassis which may be fitted into the recorder cabinet. The input connections may be permanently soldered to the ground and cathode pins of tube VT-3 in the recorder—or they may be temporarily connected by means of an octal adapter base inserted under this tube. Power is supplied from the cable and socket which formerly connected to the 6E5. The input connector of the selector-VI circuit plugs into the output connector of the VI amplifier, picking up the audio signal and pilot-bulb power circuits. A pair of leads may be run to the line-level recorder output to provide VI and earphone monitoring of output with the selector switch set to "line".

and adjustment panel for the audio circuits. A switch is provided to turn the recorder loudspeaker on or off; a low-impedance potentiometer across the output transformer secondary provides an adjustable recorder output to be fed to an external amplifier; a high-impedance potentiometer enables control of input signal level. In the present system, a cathode-follower unit installed in the receiver transmits the detector output at low-impedance to the recorder input. With the added "radio-playback" switch on the recorder set to "radio" this signal is also transmitted back to the receiver audio channel. With the switch set to "play-back", the recorder output signal is transmitted to the receiver audio channel instead. In cases where the recorder audio system is superior to that of the receiver used, transmission back to the radio may be omitted.

Although the modifications may sound a little complex, non-technical persons have experienced no difficulty in operating the device and have not

Fig. 7. Preliminary model of assembly for splicing magnetic recording paper tape. A guillotine trimmer is shown being used to trim off excess "scotch-tape" from a splice. The two pieces of tape are first squarely cut off in the modified film splicer, being held in the milled grooves in the lower clamp jaws, and a short length of "scotch-tape" is applied to the back (paper side) of the butt-jointed tape. The spliced tape is then removed to the trimmer where the two sides are trimmed successively. Such an assembly is especially useful in taking over all holding and spacing operations in the splicing process.



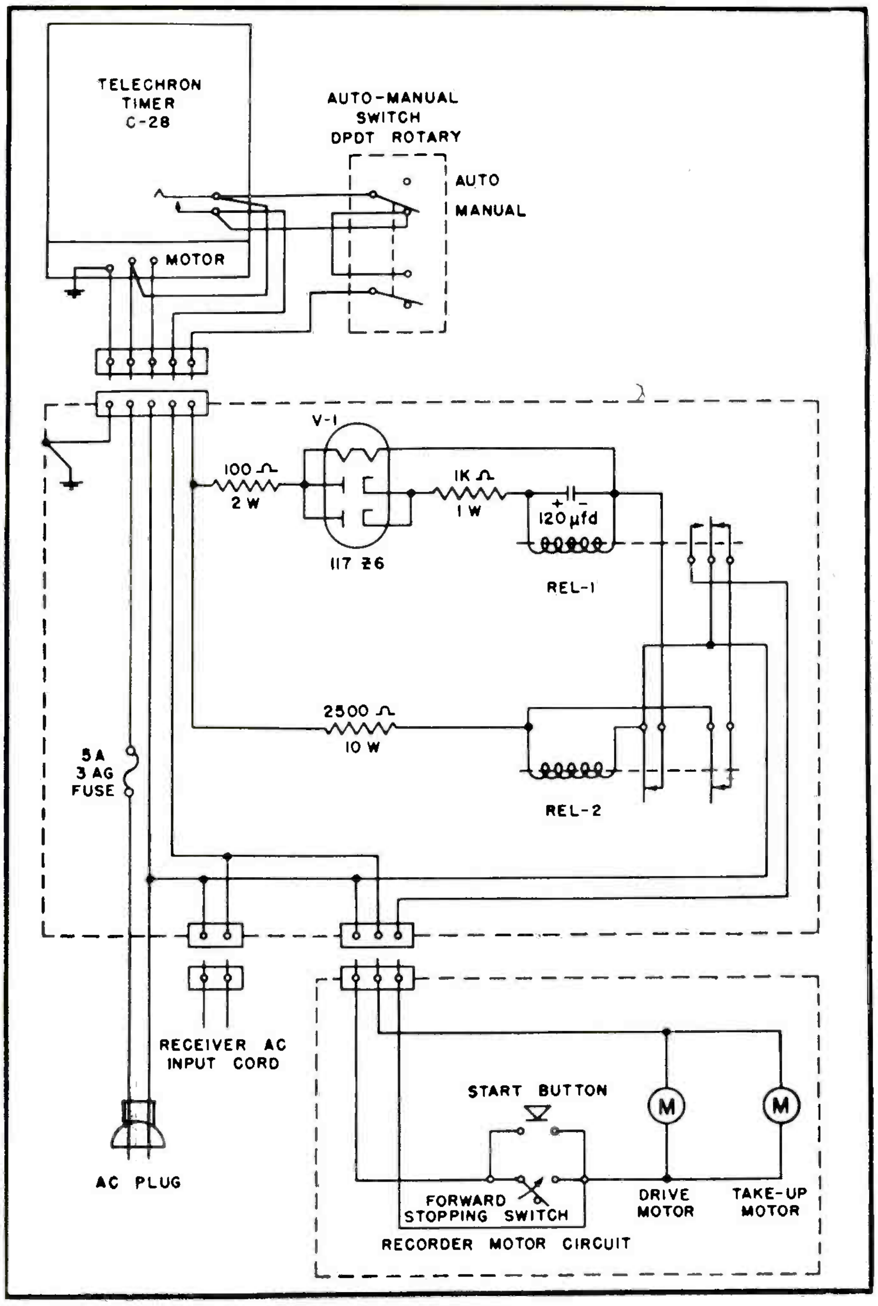


Fig. 5. Control-circuit for automatic operation of Brush BK-401 recorder. Both radio receiver and the recorder plug into the control circuit chassis. With the auto-manual switch on "manual" as shown, power is supplied to the radio and recorder continuously so they may be operated normally and independently. With the switch set to "auto", power is supplied only through the timer contacts in accordance with a preset schedule. In addition, a delayed starting pulse is supplied to the recorder motors by the relay circuit. This pulse enables the motors to pull-in the forward stopping switch. Relay 1 is a 5,000 to 8,000 ohm sensitive relay; relay 2 is a midget 110-V a-c relay.

found that the automatic features add any new problems.

Maintenance

Absence of critical adjustments makes the tape recorder as simple to maintain as to operate. No unusual electrical maintenance problems are involved and the chief requirement is to clean off any accumulations of tape coating material in the recording and erase heads.

When this material rubs off as the tape passes the magnetic heads, it tends to form a cake over part of the face of the head. The tape is then pushed away from the head, resulting in an increased air-gap length. When this occurs on the erase head, erase efficiency is decreased, allowing the previous recording to remain on the tape. When it happens on the play-record head, the signal level on playback is very low and high-frequency response is noticeably impaired. The cake may be removed by dampening with a solvent such as carbon tetrachloride, acetone, or alcohol (taking care to keep the solvent away from the tape); rubbing with a soft cloth; and, if necessary, fine polishing with a strip of crocus-cloth cut to fit into the head slots.

The coating material also tends to collect in the reverse stopping-switch guide, through which the tape passes on rewind. If enough "mud" collects, the switch will be jammed so that as the tape end passes, a-c power is kept on the supply motor, allowing it to run indefinitely at high speed. Any accumulation in this guide should be cleaned out.

Developments are under way, it should be noted, which are expected to greatly reduce and possibly eliminate this rubbing-off tendency. However, with currently available tape, these cleaning operations should be done regularly—after every two to four hours of use.

After long use, the recorder may be unable to pull the tape on forward operation. This is caused by a smoothing or "greasing" of the drive-capstan cork surface. When this occurs, the cork should be rotated against a cloth dampened with carbon tetrachloride and then against a strip of 6/0 or finer garnet paper. The sanding may be followed by a second wash to pick up the dust.

These few maintenance operations together with routine tube-checking, circuit inspection for overheated components, checking for noisy controls and switches will usually suffice to keep the recorder ready for use.

Tape Handling

As paper-tape magnetic recording becomes more widely used, techniques of tape handling which will enable realization of the full capabilities of the medium will come in for attention. An initial problem is that of splicing the tape for editing purposes. Although splices can be readily made with scissors and a roll of "scotch-tape", more convenient methods are desired for professional operations. *Figure 7* illustrates a first attempt to build a splicing unit that provides the accuracy and extra fingers a person finds useful. The assembly includes a modified film splicer, a trimmer, and a "scotch-tape" dispenser.

The tape is held, coated-side down, in a groove milled in the lower clamp jaw of the splicer by a felt-pad cemented to the upper clamp jaw. A special cutter blade is attached to each lower clamp jaw in such a way that both pieces of tape to be joined are cut on the same edge. After both pieces have been cut they are held in proper alignment for application of a short piece of ½" wide "scotch-tape". A special guillotine trimmer is used to trim off the excess "scotch-tape". A trimmer which may be operated more easily is planned for a second splicing assembly.

Improvement and production of tools such as this will be needed as will the

[*Continued on page* 48]

Design and Construction of

Practical Dividing Networks

C. G. McPROUD*

Instructions for making loudspeaker dividing networks without using elaborate laboratory equipment.

Although the subject of dividing networks has been covered with considerable thoroughness in the literature over the past few years, much of this material has made it necessary for the builder to make a number of choices as to the circuit used. Few writers on the subject have made definite recommendations, most of them providing details for the calculation of *all* types of network configurations. In addition, the builder who does not have a suitable bridge available for measurement of the required inductances is handicapped because of the difficulty in obtaining ready-wound coils for this application. It is the purpose of this article to describe one particular type of dividing network, the choice having been made on the basis of a number of listening tests. Furthermore, constructional details are furnished which make it possible to wind acceptable coils for these networks without the need for measuring the inductance values on a bridge. Provided the coils are wound exactly as described, the resulting inductance should be within 5 per cent of the required value; this may be of some aid to those who do not have access to the necessary measuring equipment.

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Network Design

Dividing networks are designed by two entirely different methods. The most common is based on the filter principle, and the characteristics of the sections resemble those of any other type of filter. The other method is based upon obtain-

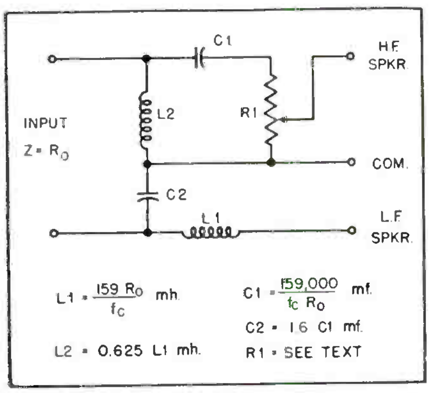
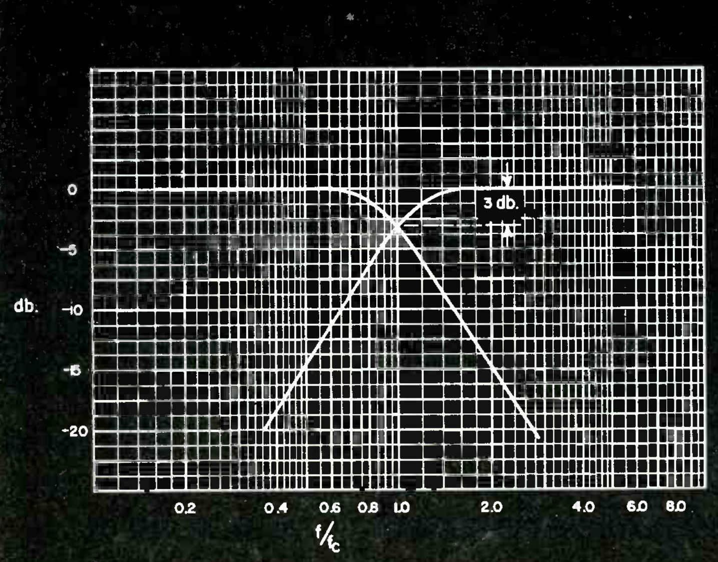


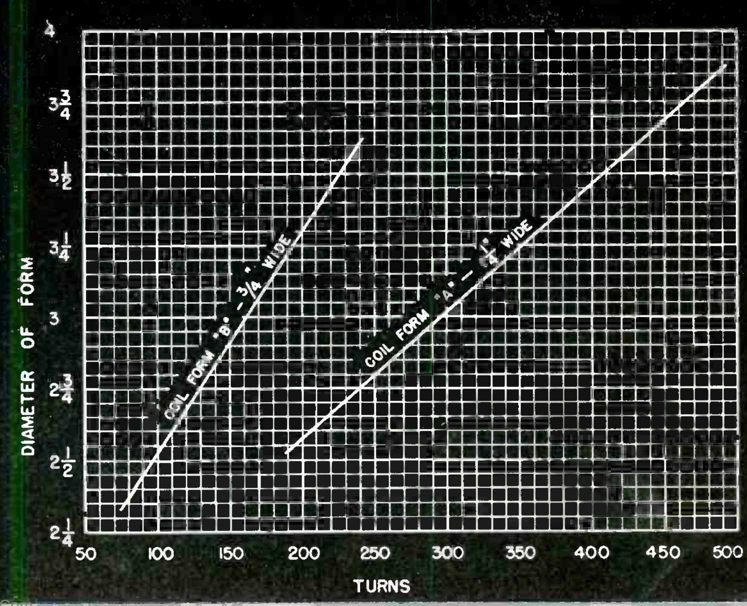
Fig. 1. Schematic of dividing network discussed in this article.

ing a constant-resistance characteristic of the sections, and while the configurations are similar, the method of calculating the values for the components differs considerably. Although the idea of maintaining a constant-resistance load on the output of an amplifier, with a consequent stabilization of the source impedance out of which the speakers work, is one which appears to be in the realm of good design, it cannot be denied that no loudspeaker offers a truly constant impedance to its source. For this reason, the filter type cannot be eliminated from consideration, since the impedances resulting from the use of this network do not vary more than 20 per cent throughout the range for which they are designed. More important, however, is the fact that critical listening tests have usually resulted in the selection of the filter network. It is, therefore, the one selected for this article. The principal advantage to be gained from the use of the constant-resistance network is one of simplification of manufacture, since both inductance elements are of the same value, and both capacitors are of the same value. For the constructor who has occasion to make but one or two, this is not an important consideration.

The next step in selection of a network type is based on the required attenuation beyond the crossover frequency. Single L-section networks provide an attenuation of approximately 12 db per octave outside the transmitted band, and if greater attenuation is desired, a full T- or pi-section is used to give a loss of about 18 db per octave. The use of properly designed high-frequency

Fig. 2 (left). Frequency-response characteristics of dividing network of Fig. 1.
Fig. 4 (right). Flange diameter required for coil forms for coils having given number of turns of No. 17 DCE wire.





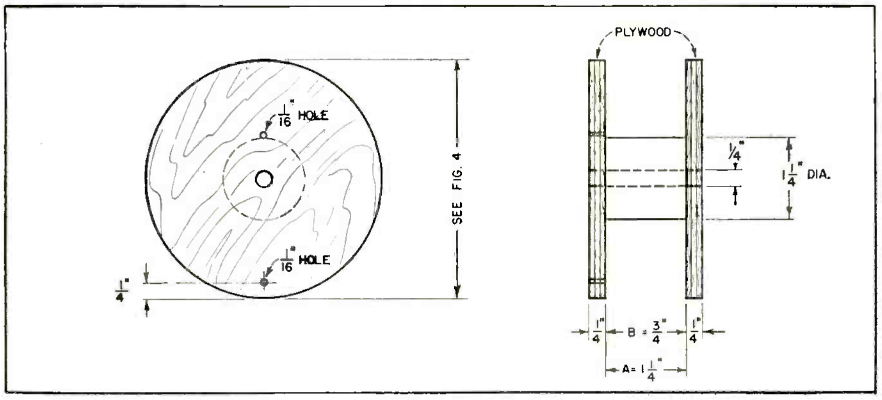


Fig. 3. Coil forms used in winding inductances in accordance with curves of Figs. 4 and 5.

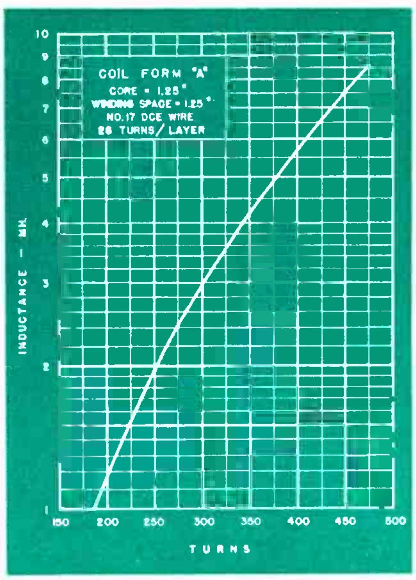
horns will aid in the attenuation of the low frequencies provided the horns are constructed so that there is a natural falling-off of output as the frequency is lowered. Thus, with a crossover at 1,000 cps, for example, the horn itself should be designed to cut off around 750 cps, so that additional attenuation is not necessary. This same consideration applies to some extent with the low-frequency speaker, although unless special speakers are used, they may supply considerable energy well above crossover. Again resorting to the results of listening tests, it is usually sufficient to employ single L-section networks, with their 12-db/octave attenuation. One other consideration in this selection is the loss due to the use of additional reactive elements. If there were no resistive component in the coils and capacitors used in networks, any number of sections could be used without any additional loss. However, with practical coils and capacitors there is always some loss, and even when relatively large wire is used in the inductances, a loss of 0.5 to 1 db must be expected through any network.

Arrangement

The final choice in the network selection is that of the arrangement of the sections. Either parallel or series connection may be employed, with differing results. The characteristic of the parallel network provides that the impedance outside the transmitted band rise to infinity, thus offering little or no damping to the speaker unit at these frequencies, even though they are not in the transmitted band for that particular speaker. The objection to this circuit may be questioned, but again basing the selection upon the all-important listening test, the series network is chosen for this discussion. The impedance of each section of the filter network falls to zero outside the transmitted band.

The final form for the network under consideration is shown in *Fig. 1*, together with the formulas for the calculation of the various components. The constructor will have selected the desired crossover frequency based upon the efficiency of either or both speakers throughout the audio band. Crossover frequencies below about 800 cps require the use of larger and longer horns, if the multi-cellular type of horn is used. If the low-frequency speaker is of the folded-horn type, the transmission may be limited to as low as 500 cps, requiring the lower crossover. In general, it is usually considered advisable to make the crossover frequency as low as possible in order to get as much of the mid-range frequencies out of the high-frequency unit, particularly if a metal-diaphragm unit is being used with a multi-cellular horn. Very satisfactory results have been obtained with crossovers ranging from 800 to 1,000 cps, while if the space permits, still better results can usually be obtained from a crossover at 500 cps. Some high-frequency units are designed for crossovers of 1,200, 2,000, and 3,000 cps, and while the output in the upper ranges is improved by the use of a two-way system with the higher crossover, greater intelligibility and naturalness is obtained with the middle speech-frequency range (1,000 to 2,000 cps) supplied

Fig. 5. Turns necessary to obtain various inductance values on coil form "A."



by the metal-diaphragm unit. These factors must all be considered in making any choice.

Configuration

The network configuration arrived at through these choices is shown in *Fig. 1*, which is an L-section series network. It is designed to feed two speakers of the same impedance, at any crossover frequency desired, with the components calculated against these parameters. If the high-frequency unit is of a different impedance from that of the low-frequency speaker, the additional efficiency of this unit may be used to permit the matching to the impedance of the low-frequency speaker by the proper choice of value for R_1, as will be described later.

Since odd values may often be obtained from the formulas for C_1 and C_2, a simple artifice may make it possible to use standard, commonly available values for the capacitances. Suppose, for example, that the formulas yield values of 14.2 and 22.7 μf for the two capacitors for a crossover frequency of 800 cps. Capacitors having even values can be substituted for the calculated units, maintaining the same ratio between the two values, so that 15.0 and 24.0 are usable, resulting in a change of the crossover frequency from 800 cps to 845 cps. This figure is then used for calculations for the inductances, and little change in the performance of the system should be observed. The advantage of such a change lies in the use of capacitor values more easily obtained, without the necessity of building up the required total capacitance with a number of small units.

It is considered good practice to be able to radiate power from a loudspeaker to at least half an octave beyond the crossover point in order to provide satisfactory operation in the crossover region. Thus, if the crossover is to take place at 1,000 cps, output from the high-frequency unit is necessary down to 1.5 (0.5 x 1,000) or 750 cps. Any shift in the predetermined crossover frequency should take this into account.

Figure 2 shows the transmission curves for both sections of the network of *Fig. 1*. It will be noted that at crossover each circuit has a loss of 3 db, making the total output power at crossover equal to the input power. This may necessitate some compensation for the increased efficiency of the high-frequency unit, and R_1 is provided for this purpose. For nominal differences in efficiency of the two speakers, of the order of 6 db, the value of R_1 may be chosen at 1.5 times the impedance of the network or speakers, assuming that both speakers are of the same impedance. This will allow a reasonable margin for adjustment of the relative level fed to the high-frequency unit, without greatly disturbing the load impedance on the h-f branch of the network. For more critical applications, the resistor R_1 can be replaced by an L-pad, or with fixed resistors calculated from the desired attenuation and the impedance of the speaker.

When the high-frequency unit has an impedance differing materially from that of the low-frequency speaker, as is often the case when 8-ohm speakers are used for the low-frequency unit and 16-ohm high-frequency units are employed on a horn, the difference in efficiency permits the adjustment of impedances by the use of a simple shunt resistor at R_1 without the use of a tap. Thus, if the efficiency differs by 6 db, the 16-ohm unit should be shunted by 16 ohms, and fed from an 8-ohm network, when used with the 8-ohm low-frequency speaker.

Coil Construction

The principal objective of this article is to simplify coil construction to the point where reasonably good results may be obtained without resorting to a bridge for the measurement of inductance values. With this in mind, therefore, the coil forms shown in *Fig. 3* are used as the standards for these data, two sizes being provided to take care of a wide range of inductances while maintaining a reasonable form factor for the coils. Therefore, it is recommended that coil form "A", with a winding space of 1¼" be used for coils ranging from 2.0 to 8.0 mh, and that coil form "B", with a winding space of ¾" be used for coils of 0.5 to 2.0 mh. Assuming that the constructor has determined the inductance values required for a network and selected the form most suitable for winding the coils, he may determine the maximum outside diameter for the flanges from *Fig. 4*. These values, together with the information provided in *Figs. 5* and *6* are predicated upon the use of No. 17 DCE wire. This may appear to be an unusual size, but it is probable that anyone interested in two-way speaker systems will have, or at least will have access to, a 6-volt field coil from a Western Electric 555 unit which is wound with wire of this size. Although having made some twenty or thirty sets of coils for dividing networks, the writer has never yet used any wire other than that from these field coils.

Returning to *Fig. 4*, if the flanges for the coil forms are cut to the dimensions shown, they will allow suitable winding space with about ¼" overlap, thus giving coils of uniform appearance. The curves of *Figs. 5* and *6* may be read directly to determine the number of turns of wire to be wound on each coil form for the required inductance value. Note that twenty-three turns are wound per layer on the large form, and that thirteen turns are wound on each layer of the

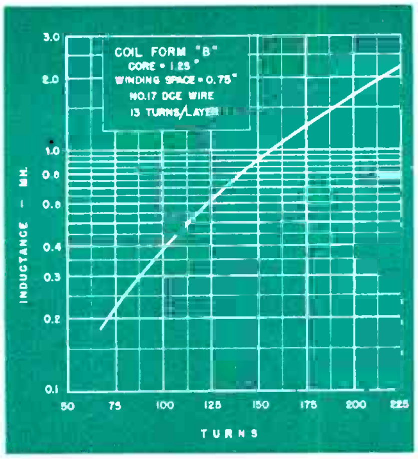


Fig. 6. Turns necessary to obtain various inductance values on coil form "B".

small form. If wire of other size must be used, comparable values for the number of turns required will be obtained if the number of turns per layer is held to either of these values. This will, however, require a recalculation for the width and diameter of the forms. The curve of *Fig. 5* covers inductance values from 1.0 to 8.0 mh, while the curve for *Fig. 6* covers values from 0.2 to 2.0 mh. When carefully layer-wound by hand or on a lathe, Q values of the order of 20 to 25 should be obtained, which are adequate for dividing network coils.

It goes without saying that no iron should be used in the construction of these coil forms but that all parts of the form are of wood or plastic. It is recommended that they be made by gluing the flanges to the core. A ¼" hole through the center may be used for mounting the forms by means of large brass wood screws. Some constructors prefer to wind coils on a demountable form, and after winding to remove the coil from the form and tape it up. This is somewhat of a refinement, and when only one set of coils is required, it is simpler to wind the form with the required number of turns and leave the winding in place. The start and finish of the winding may be brought out through holes in the flange and attached to suitable terminals. Two or three coats of lacquer will furnish a protective coating for the coils, but if they are to be used in humid climates, they should be baked out and impregnated with varnish.

After winding the coils, the network should be assembled on a wooden base, making sure that the two coils are not mounted closer than six inches from each other, and that their axes are at right angles. If it is desired to enclose the entire network in some fashion, a wooden box is recommended; the use of metal in proximity to the coils tends to reduce the Q, and may affect the overall performance.

Tests

If an audio oscillator and gain set is available, the constructor will wish to make measurements of the transmission characteristics through each section of the network, the other section being suitably terminated. If such equipment is not available, it is possible to make a reasonable adjustment solely by means of the ear, which is the ultimate instrument for which the speaker system is designed. For the constructor who makes several speaker systems, the test unit shown in *Fig. 7* will prove an aid in balancing and phasing.

The two-pole, three-position switch permits the feeding of either speaker independently, or both together. When either section is cut off, the circuit is terminated properly. The 1-db/step pot provides for the attenuation of the high frequency unit for determination of the required amount which must be introduced by the resistor R_1 in *Fig. 1*. The DPDT switch in the leads to the high-frequency unit provides for reversing the

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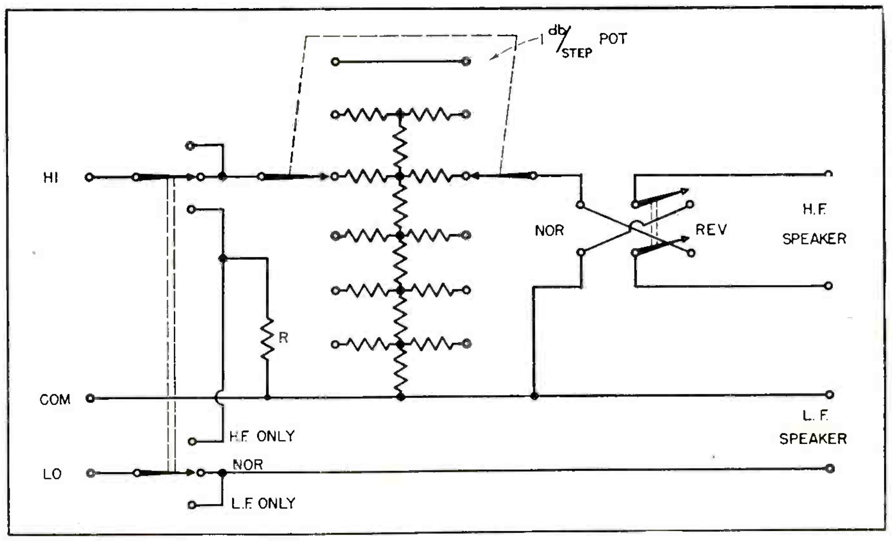
Fig. 7. Test unit for simplifying the balancing and phasing of l-f and h-f speakers.



Cathode Phase Inverter Design

C. W. VADERSEN*

General design equations and an investigation of the degenerative properties of the cathode phase inverter are presented.

PART I

THE CATHODE PHASE INVERTER has found wide application in recent years and promises to lend itself to more uses in the future. In this article an attempt will be made to make a more detailed analysis of its operation and design, than has hitherto been published. It will be shown that the circuit can be designed in a variety of forms, some of which exhibit such desirable features as economy of components, stability of balance and over-all simplicity. In addition, it will be demonstrated that degeneration only exists in one side of the amplifier with a theoretical maximum amount of 6 db. The use of an output transformer with unbalanced high windings, to obtain a phase inverting power output stage, is also discussed. Design equations are developed in both the μ, R_P form and the transconductance form. The latter are sufficiently accurate for triode designs, if the indicated correction for load resistance is applied to the tube transconductance.

Circuit Analysis

Figures 1 and *2* show the basic circuit and its equivalent for the cathode phase inverter. The operation of this circuit is based on the fact that the signal current i_1 is larger than i_2, leaving a residual voltage across R_K which is 180° out of phase with e_{g1}. With this in mind it is simple to set up the proper equivalent plate circuits (based on linear operation) and solve for i_1 and i_2 by application of Kirchhoff's laws. This procedure will yield the following results:

$$i_1 = \frac{\mu_1 e_s [R_{P2}+R_{L2}+R_K(\mu_2+1)]}{D} \qquad (1)$$

$$i_2 = \frac{\mu_1 e_s R_K(\mu_2+1)}{D} \qquad (2)$$

$$D = [R_{P1}+R_{L1}+R_K(\mu_1+1)] \times [R_{P2}+R_{L2}+R_K(\mu_2+1)] - R_K^2(\mu_1+1)(\mu_2+1) \qquad (3)$$

The condition for balanced output voltages can be written as

$$i_1 R_{L1} = i_2 R_{L2} \qquad (4)$$

or

$$R_{L1} = \frac{i_2}{i_1} R_{L2} \qquad (5)$$

By substituting equations 1 and 2 in equation 5, the necessary relation between R_{L1} and R_{L2} is established for balance.

$$R_{L1} = \frac{(\mu_2+1)R_K R_{L2}}{[R_{P2}+R_{L2}+R_K(\mu_2+1)]} \qquad (6)$$

Several interesting things are revealed by inspection of equation 6. The most obvious one is that the tube V_1 has no effect whatsoever on the balance. However, it does affect the gain, which will be demonstrated later. The second thing to be noted is that as the product $R_K(\mu_2+1)$ increases, the value of R_{L1} approaches that of R_{L2}.

The expression for R_{L1} given by equation 6 is useful for triodes only when μ and R_P are known. For pentodes, the transconductance form is applicable.

$$R_{L1} = \frac{gm_2 R_K R_{L2}}{1+gm_2 R_K} \qquad (7)$$

When using equation 7 it should be borne in mind that the effective value of transconductance should be used for gm_2. If this is done, 7 may also be used for triodes. The effective gm can be calculated from the relation

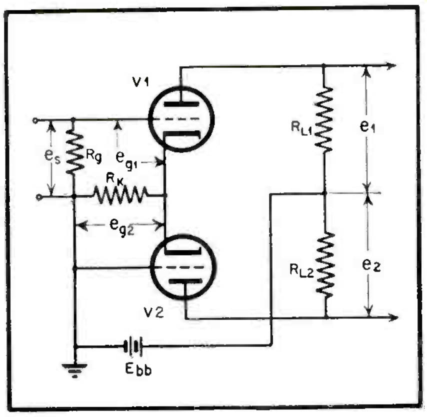


Fig. 1. Basic cathode phase-inverter circuit.

$$gm_2 = gm_0 \frac{R_{P2}}{R_{P2}+R_{L2}} \qquad (8)$$

or can be found on the published plate family, using a load line determined by R_{L2}. gm_0 is the transconductance of V_2 with $R_{L2} = 0$, but at the same operating voltages as obtained with R_{L2} equal to the assigned value.

A measure of the output balance can be found in the ratio e_1/e_2.

$$\frac{e_1}{e_2} = \frac{i_1 R_{L1}}{i_2 R_{L2}} = \frac{[R_{P2}+R_{L2}+R_K(\mu_2+1)]}{R_K(\mu_2+1)} \times \frac{R_{L1}}{R_{L2}} = \beta \qquad (9)$$

In the transconductance form,

$$\beta = \frac{1+gm_2 R_K}{gm_2 R_K} \times \frac{R_{L1}}{R_{L2}} \qquad (10)$$

From these relations the change in balance, due to changes in R_{L1} and R_{L2}, can be seen by inspection. The effect of changes in μ, R_K, R_P and gm can be demonstrated by simply writing the derivatives with respect to these quantities as follows:

$$\frac{d\beta}{d\mu_2} = -\frac{R_{P2}+R_{L2}}{R_K(\mu_2+1)^2} \times \frac{R_{L1}}{R_{L2}} \qquad (11)$$

$$\frac{d\beta}{dR_K} = -\frac{R_{P2}+R_{L2}}{R_{K^2}(\mu_2+1)} \times \frac{R_{L1}}{R_{L2}} \qquad (12)$$

$$\frac{d\beta}{dR_{P2}} = \frac{1}{R_K(\mu_2+1)} \times \frac{R_{L1}}{R_{L2}} \qquad (13)$$

$$\frac{d\beta}{dgm_2} = -\frac{1}{gm_2^2 R_K} \times \frac{R_{L1}}{R_{L2}} \qquad (14)$$

The most interesting thing shown by these relations is that by having either of the quantities $R_K(\mu_2+1)$ or $gm_2 R_K$ large, the circuit can be made to have considerable stability in balance of output voltages. This will be illustrated by specific examples later on. The negative signs merely indicate that as the values of μ, R_K and gm increase, the ratio e_1/e_2 decreases.

Voltage Gain

Another useful relation in design procedure would be an expression for the voltage gain. This can be written from equations 2 and 3 as:

$$V.G. = \frac{e_2}{e_s} = \frac{i_2 R_{L2}}{e_s} = \frac{\mu_1(\mu_2+1)R_K R_{L2}}{D} \qquad (15)$$

By inspection of equation 15 it can be seen that the gain is dependent on the tube constants of both V_1 and V_2 although the output balance is only affected by V_2 as shown by equations 6 and 9. The equivalent transconductance form is

$$V.G. = \frac{gm_1 gm_2 R_K R_{L2}}{1+R_K(gm_1+gm_2)} \qquad (16)$$

In most applications the tubes V_1 and V_2 are the same type so that the expressions for voltage gain reduce to

$$V.G. = [\mu(\mu+1)R_K R_{L2}]/[(R_P+R_{L1})(R_P+R_{L2})+R_K(\mu+1)(2R_P+R_{L1}+R_{L2})] \qquad (17)$$

and

$$V.G. = \frac{gm^2 R_K R_{L2}}{1+2\,gm\,R_K} \qquad (18)$$

For triodes, equation 16 must be used when designing by transconductance because the effective values of gm_1 and gm_2 will not be equal when R_{L1} and R_{L2} are unbalanced. Equation 18 is sufficiently accurate for pentodes, however.

Effect of Output Loading

Since R_{L1} and R_{L2} represent effective a-c load resistances in the equations, the shunting effect of the following grid resistors

*Bell Telephone Laboratories.

should be taken into account, if they are less than about $10\,R_{L2}$. A convenient way to do this is to substitute the parallel value of R_{g2} and the chosen plate resistor for V_2 for R_{L2} in either equation 6 or 7 and 8. The value of R_{L1} thus found represents the parallel combination of R_{g1} and the actual plate resistance R'_{L1}. R'_{L1} is then found from the relation

$$R'_L = \frac{R_{L1}\,R_{g1}}{R_{g1} - R_{L1}} \qquad (19)$$

In doing this it should be remembered that when determining R_K, for the quiescent bias, the actual values of the plate resistors are used and not the parallel values with R_{g1} and R_{g2}. The value of gm_2 in equation 7 is calculated by use of the parallel value of R'_{L2} and R_{g2}.

The stage can be designed so that its output balance will only be disturbed by a small predetermined amount due to loading, in the following way. Assuming R_{L1} and R_{L2} are shunted by equal resistances that approach zero, inspection of equation 9 indicates that β will approach a maximum value of

$$\beta_m = \frac{R_{P2}+R_K(\mu_2+1)}{R_K(\mu_2+1)} \qquad (20)$$

From this the value of R_K is

$$R_K = \frac{R_{P2}}{(\mu_2+1)\,(\beta_m-1)} \qquad (21)$$

By substituting this value of R_K in equation 9 and solving for the ratio $\frac{R_{L1}}{R_{L2}}$ yields

$$\frac{R_{L1}}{R_{L2}} = k = \frac{\beta R_{P2}}{\beta_m(R_{P2}+R_{L2}) - R_{L2}} \qquad (22)$$

where R_{L1} and R_{L2} are actual plate resistances.

With equations 21 and 22 the cricuit can be designed to have any initial balance β, including unity, and have a maximum unbalance of β_m, as the grid resistors of the following stage approach zero ohms. Besides affording a circuit design for specified stability limits, these relations might also be used in the case where the inverter

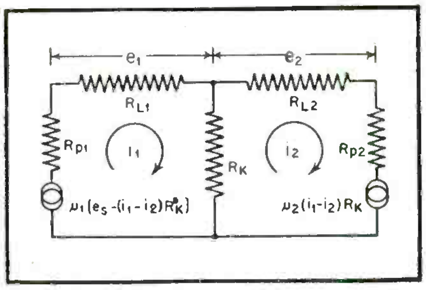


Fig. 2. Equivalent (linear) plate circuit of cathode phase inverter.

stage is driving grids that draw current. Furthermore if $\beta = \beta_m$, the unbalance is a predetermined constant amount. It should be kept in mind that equations 19 to 22 inclusive are valid only when the plate resistors of the inverter stage are effectively in parallel with the following grid resistors. The equivalent transconductance forms of 21 and 22 are:

$$R_K = \frac{1}{g_{mo}(\beta_m-1)} \qquad (23)$$

and

$$k = \frac{g_{m2}\beta}{g_{m2}+g_{mo}(\beta_m-1)} \qquad (24)$$

where g_{mo} is the tube transconductance at the chosen operating voltages and g_{m2} is the effective transconductance with the load resistance, as given by equation 8.

Negative Feedback

In discussing the amount of feedback present in the circuit it is convenient to consider one half of the output at a time. The voltage gain can be written as

$$V.G._1 = \frac{e_1}{e_s} = \frac{i_1 R_{L1}}{e_s} \qquad (25)$$

$$= \frac{\mu_1[R_{P2}+R_{L2}+R_K(\mu_2+1)]R_{L1}}{D}$$

which is the same as equation 15 when the circuit is balanced. Now if $R_K = 0$ there would be no feedback and

$$V.G._0 = \frac{\mu_1 R_{L1}}{R_{P1}+R_{L1}} \qquad (26)$$

If the feedback is expressed as the ratio of these gains,

$$F_1 = \frac{V.G._0}{V.G._1} \qquad (27)$$

$$= \frac{D}{(R_{P1}+R_{L1})\,[R_{P2}+R_{L2}+R_K(\mu_2+1)]}$$

By assuming the tube parameters on both sides of the circuit are equal, equation 27 reduces to

$$F_1 = 1 + \frac{R_K(\mu+1)\,(R_P+R_{L2})}{(R_P+R_{L1})(R_P+R_{L2})+R_K(\mu+1)\,(R_P+R_{L1})} \qquad (28)$$

The transconductance form is

$$F_1 = \frac{1+R_K(gm_1+gm_2)}{1+gm_2 R_K} \qquad (29)$$

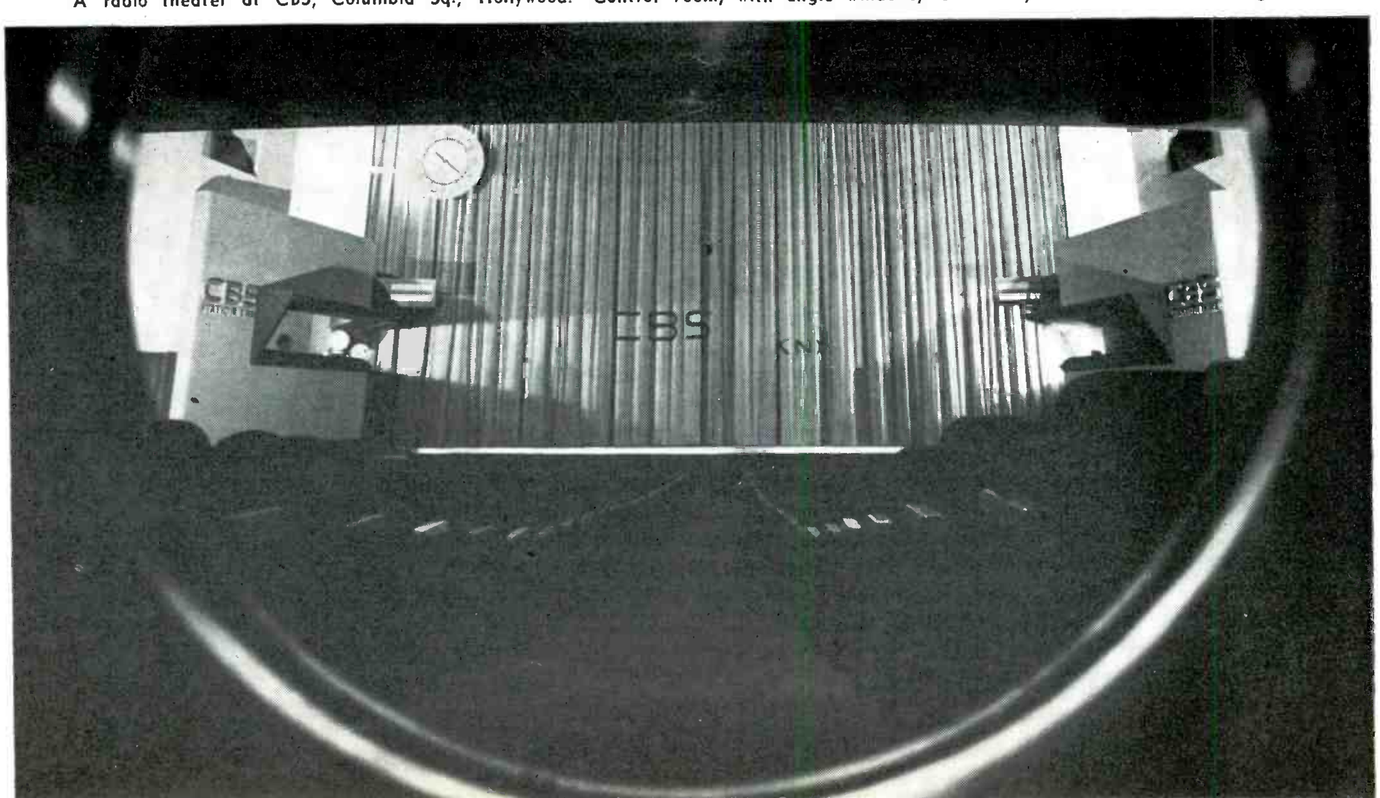
Equation 29 is valid for different as well as similar tubes in the two halves of the circuit. The transconductances are effective values, as in all other equations of this form developed in this discussion.

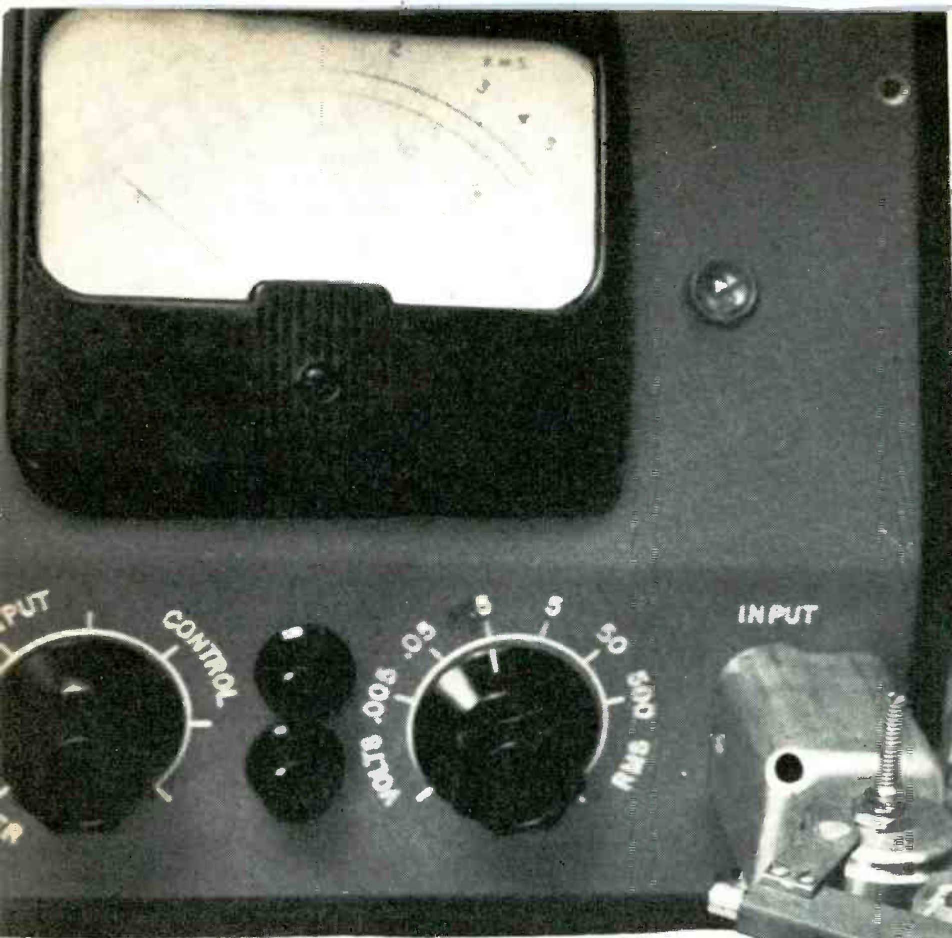
By inspection of equations 28 and 29, it is obvious that the feedback on V_1 is a function of R_K. When $R_K = 0$, $F_1 = 1$. As R_K approaches infinity, $R_{L1} = R_{L2}$ and the feedback ratio as given by 28 approaches a maximum value of 2. In equation 29, F_1 aproaches the value of $1+gm_1/gm_2$ as $R_K \to \infty$. If the tubes are identical $gm_1 = gm_2$ because $R_{L1} = R_{L2}$ and F_1 approaches 2, as in equation 28. It is interesting to note that the feedback is negative because $F_1 > 1$, and that it cannot exceed 6 db, no matter how large R_K becomes. This means that the voltage gain of the inverter stage will never be less than half that which would be obtained with the same tube and operating conditions in a straight amplifier with no feedback.

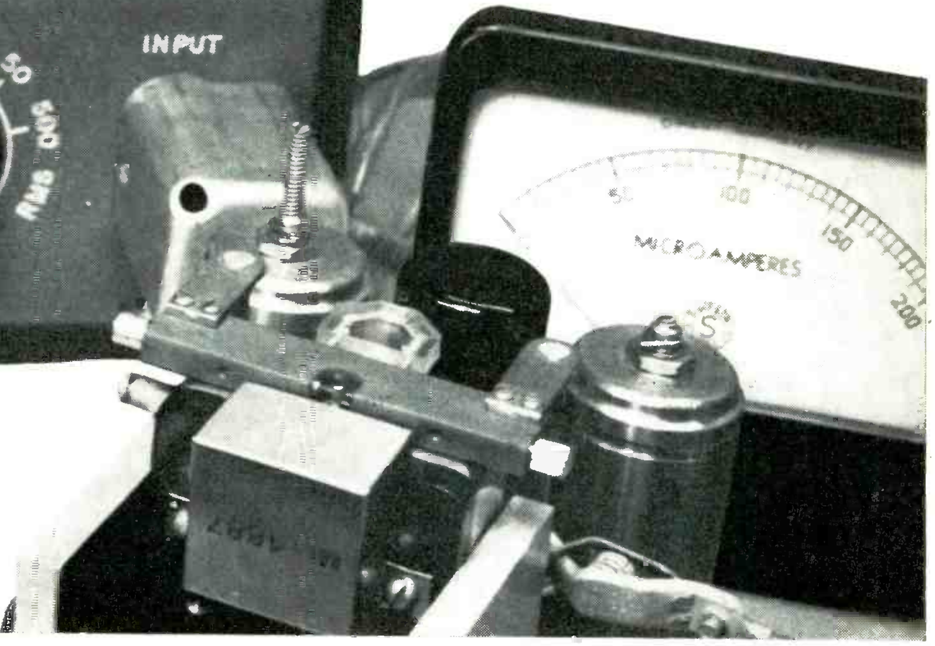
The feedback conditions for V_2 can be investigated in a similar manner. However, it must be kept in mind that V_2 is operating

[*Continued on page* 46]

A radio theater at CBS, Columbia Sq., Hollywood. Control room, with angle windows, is at left; observation room at right.







PART II

Panel view of FM calibrator unit. The sloping panel facilitates accurate reading of the meter. The jig set-up shown below is used when the instrument is employed for testing in the laboratory.

FM Calibrator For Disc Recording Heads

RALPH A. SCHLEGEL*

This is the second of two articles describing the design and construction of a successful FM calibrator.

THE REQUISITES OF AN FM CALIBRATOR for testing, measuring and maintenance of recording heads are that it be mechanically rugged, compact in design, electrically stable and consistent in performance.

In order to meet these conditions, the chassis for the oscillator-discriminator unit is machined from a block of ½″ dural, 1-9/32″ wide and 3-9/16″ long. A chassis cover plate ⅛″ thick of the same material also is required.

Lay out all dimensions on the chassis block, drill holes for the chassis cover plate and tube socket mounting holes to the depths indicated in *Fig. 1a*, tapping for 0-80 machine screws with a bottoming tap. Next, drill the coil shield mounting holes, using a #52 drill for 0-80 screw clearance and counterbore to a depth of ⅜″ from the base of the chassis for filister head screws. The holes for the coil leads, push-pull capacitor, and ground wires are then drilled in the top and sides of the chassis to a depth of approximately ¼″. The coil mounting and cable holes are also drilled to a depth of ¼″ in the top and ends of the chassis block. The tube socket hole is drilled deep enough to provide a ⅛″ side wall, one inch in diameter. The bottom of the chassis is then milled out to a depth of ⅜″ with a one-inch end mill cutter. If you don't have an old milling machine kicking around under the bench, a drill press and bench vise will do the job satisfactorily. Use plenty of kerosene on all of the drilling and tapping operations.

The reader may wonder why the drilling procedure is described in such detail. However, 0-80 screw taps are fragile tools and one's disposition becomes somewhat ruffled if the tapping of small holes is left as the last operation, only to hear the familiar "click" as the tap snaps off even with the surface of the work. Following the milling, the hole for the connecting cable is filed to an oval shape and the coil mounting holes are countersunk.

The cover plate, *Fig. 1b*, is firmly attached to the chassis so that the ends of the block and plate may be shaped to a 41/64″ radius. The shaped block is placed on a disc or belt sander using fine emery paper to finish and grain the sides of the chassis. (This is known as the "professional touch," in the better shops.)

The coil shields are cut on a lathe from 1″ inside diameter brass tubing 2-1/16″ long, and are 2-1/16″ long. The bottoms of the shields are drilled and tapped, matching the chassis mount-

*WOR Recording Studios, New York City.

ing holes, to receive the 0-80 machine screws which hold the shields securely to the chassis. Brass caps for the shields are turned from 1-1/8" diameter solid brass rod so as to give a tight fit into the top end of the shield cases. A 1/4" hole is drilled through the brass cap to pass the iron-core tuning control.

The tube socket mounting ring can be made in several ways, using a 1-1/4" outside diameter, 1/8" wall brass tubing 1/4" long, and drilling two #52 holes for 0-80 screw clearance longitudinally through the walls or by using 1-1/4" diameter solid brass rod and turning the inside to leave a 1/32" thick wall. The latter method is preferable as it helps to reduce shunt capacitance effects on tube-socket leads. It will be necessary to turn down the lower portion of the tube socket so that it will fit into the one-inch chassis hole. Two holes are also drilled and countersunk for 0-80 flat head machine screws through the tube socket between base pins. These holes must line up with the socket mounting holes on the chassis.

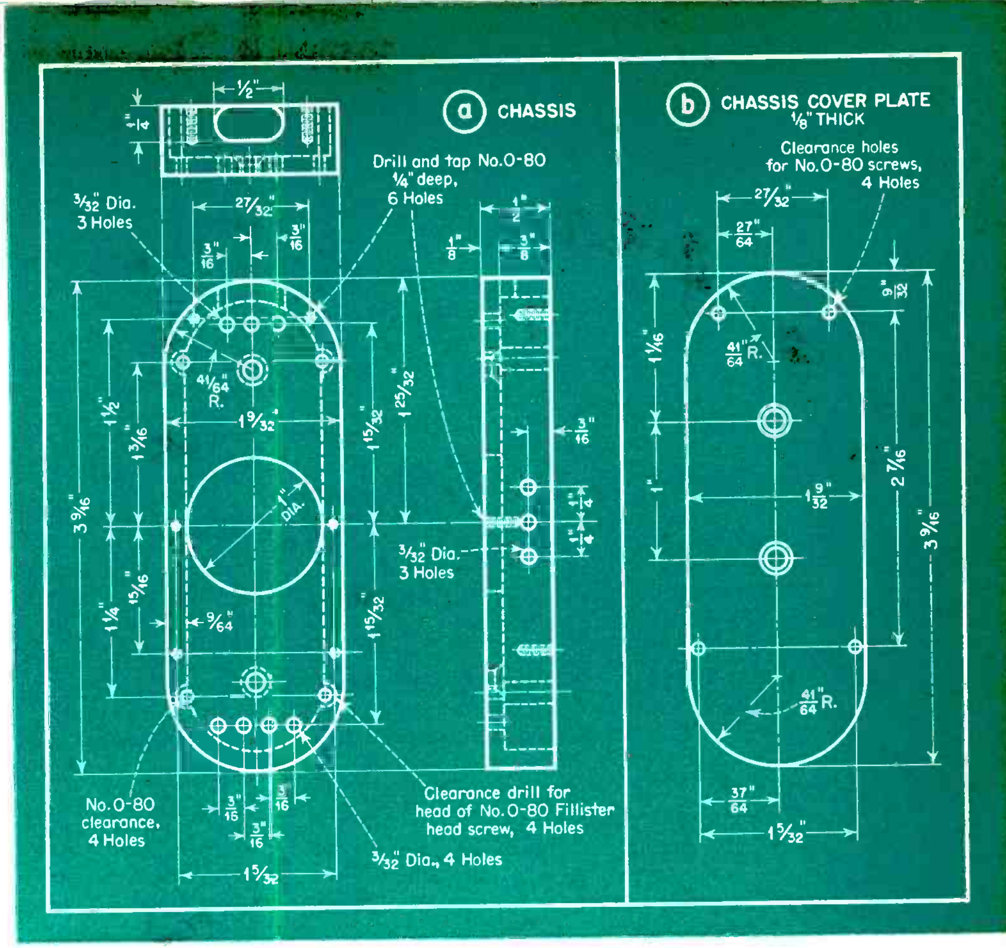


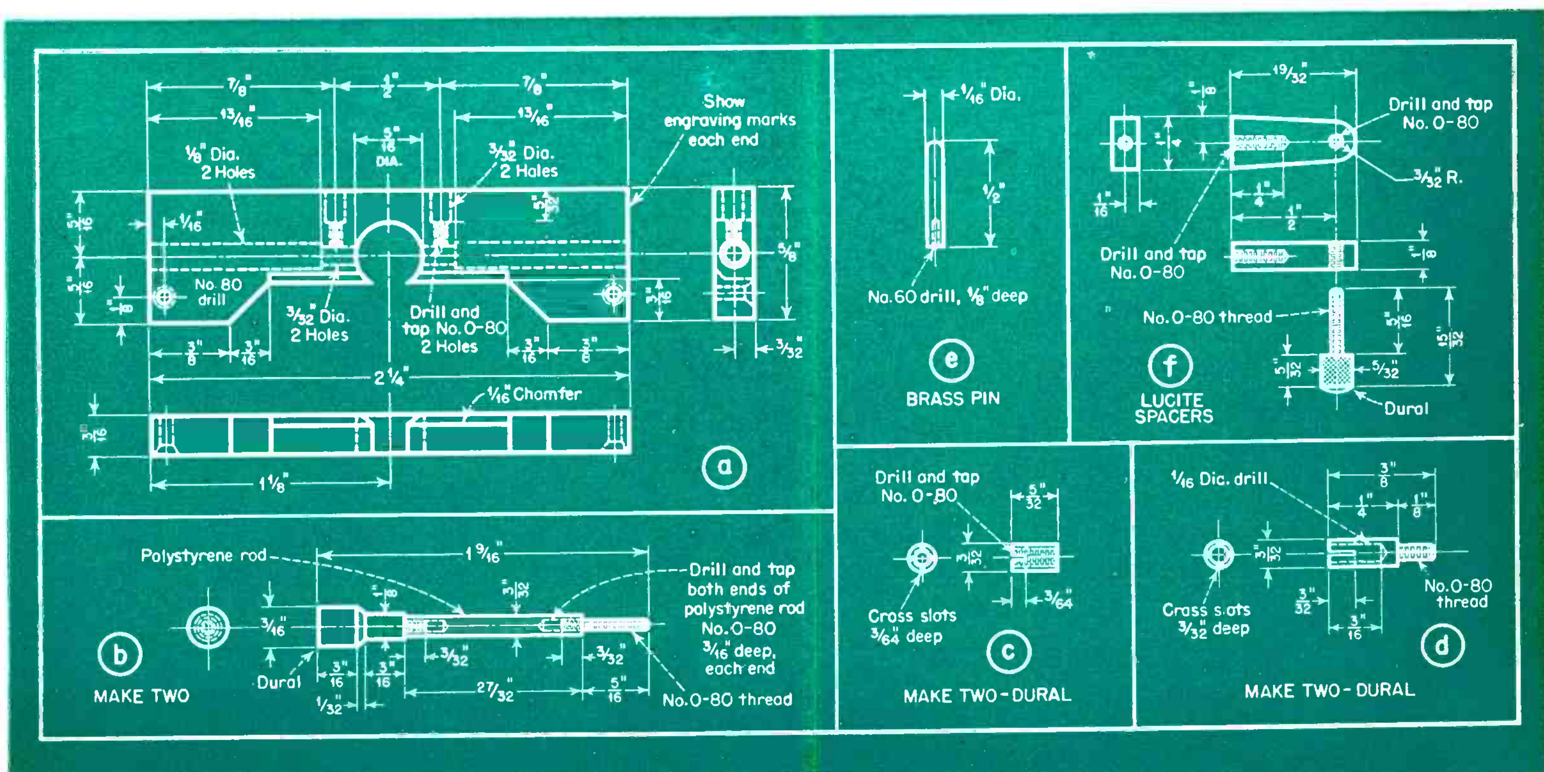
Fig. 1. In (a), constructional details for the chassis block and (b) dimensions and layout of the chassis cover plate. See text.

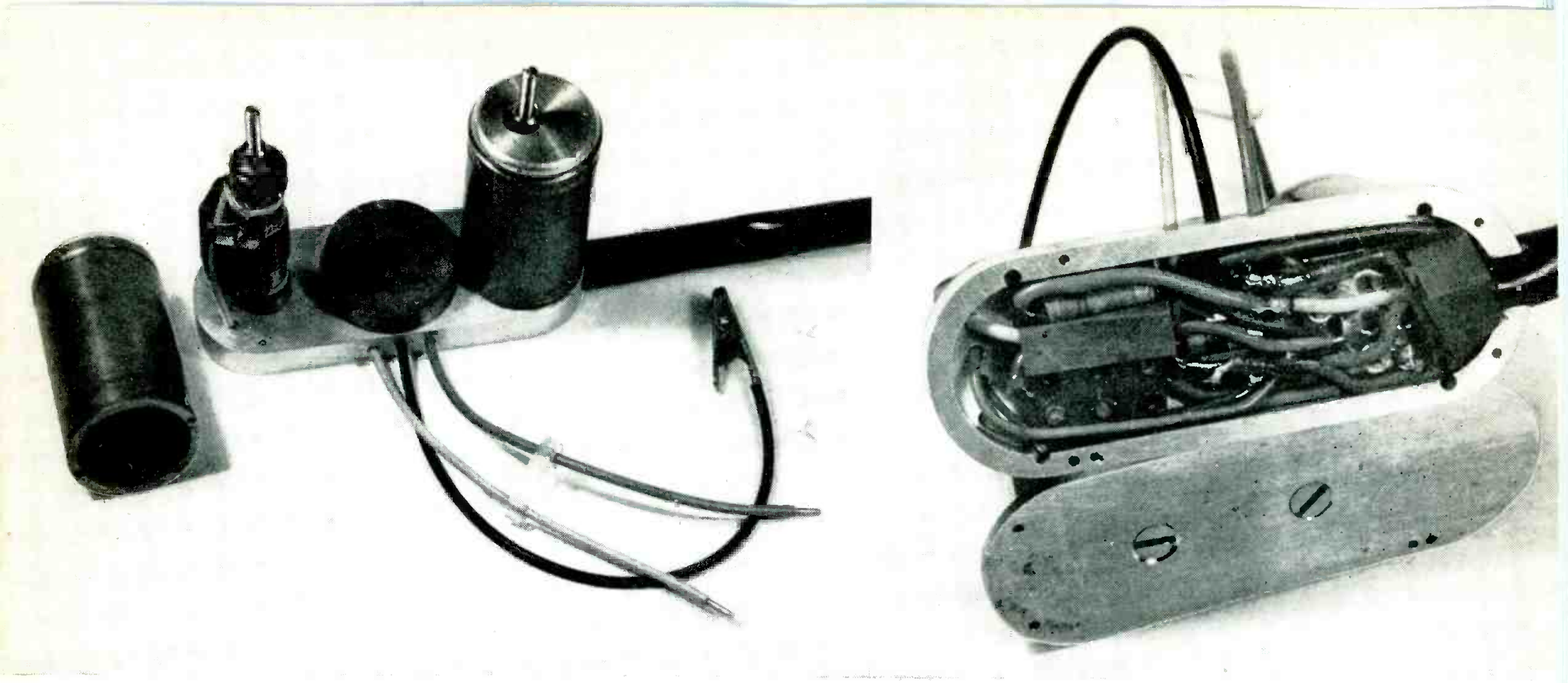
Oscillator-Discriminator Coils

The oscillator-discriminator coils are wound with #20 enameled wire on 1/2" diameter coil forms. These can be obtained by salvaging suitable permeability-tuned i-f or r-f coils, using a 3/8" diameter tuning slug. Experience indicates it will be necessary to ream the tuning slug hole to permit smooth movement of the powdered-iron core in the form. The oscillator coil is wound with 23 turns of #20 enameled wire, tapped at the eighth turn from the bottom. If a threaded form is not available, the turns may be spaced with #22 enameled wire which is removed after spacing out the turns, and the coil is then coated with low-loss coil cement. The discriminator coil is wound with 16 turns of #20 enameled wire, spaced in a similar manner and coated with cement. The oscillator grid leak, grid capacitor and 100-μμf blocking capacitor may be mounted on the oscillator coil form. The 27-μμf and 100-μμf discriminator capacitors may be similarly mounted on the discriminator coil form.

The unit is now ready for assembly. The tube socket and brass supporting ring are mounted on the chassis with two flat-head 0-80 machine screws. Coil forms are also fastened with flat-head screws. Leads to the coils and capacitors are brought through the chassis and soldered to the lugs on the coils. The coil shields are then placed in position

Fig. 2. Mechanical design and constructional details for the capacitor- plate assembly. This is essentially the same type as that used by many recording head manufacturers and service organizations.



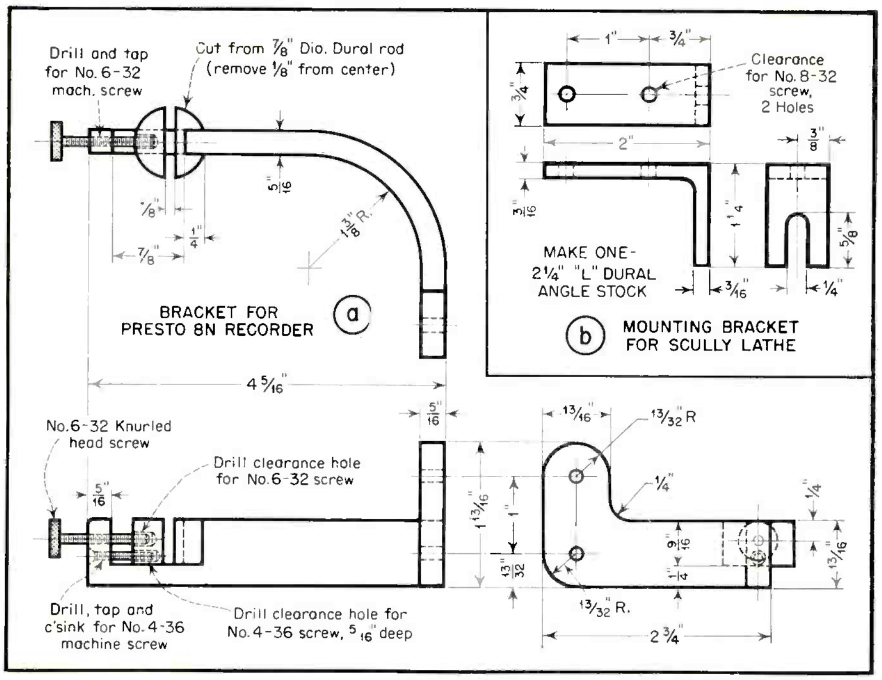


Oscillator-discriminator unit and (right) bottom view with cover removed.

and secured to the chassis with 0-80 filister-head machine screws. The two flexible leads from the capacitor-plate assembly, and the ground lead from the cutter case are made of 0.004" steel piano wire, covered with vinyl tubing. Two ¾"-long lucite spacers, drilled to clear the tubing on ½" centers, are threaded on these leads before soldering down.

The cable connecting the oscillator-discriminator unit to the audio amplifier is made of a three-foot length of 3/16" O.D. coaxial line with the insulation covering the shielding removed. Three lengths of flexible hook-up wire are laid along the coax, then all four wires are loosely wrapped with lacing cord over the entire length and covered with a length of flexible oval plastic tubing. The common B— and ground circuit is carried on the shield of the coax. If the calibrator is to be used on recording machines, care must be taken to see that the shield and *electrical* ground are insulated from the chassis of the oscillator-discriminator unit, thus permitting the chassis to obtain its ground through the recording machine, and the circuit ground wire which is clipped to the cutter head case. When the recording machine is not at ground potential, it will be necessary to ground either the machine or B— terminal at the power supply, whichever gives the best results. Failure to observe proper grounding of the equipment may lead to ground loops and unstable operation of the calibrator. If the unit is intended for use solely on the test bench in the laboratory it will be necessary to ground the chassis to the shield of the coaxial line.

Fig. 3. Mounting brackets for two types of recording lathes. For Presto 6D and Allied recorders, it will be necessary to drill and tap a 6-32 hole in the overhead carriage and make a simple "S" bracket to mount the calibrator.



Capacitor Plate Assemblies

Several types of push-pull capacitor-plate assemblies have been shown in previous articles.[1,2] While perhaps satisfactory for the laboratory experimenter, they leave much to be desired from the standpoint of adjustability and it is difficult to duplicate measurements with them. The capacitor-plate assembly now being used by several recording head manufacturers and service organizations is essentially the same as that illustrated in Part 1.* Its mechanical construction and components are shown in *Fig. 2.*

The plate holding the push-pull capacitor plates is made from a piece of 3/16" brown bakelite, ⅝" wide and 2¼" long. A 5/16" diameter hole for the cutter stylus is drilled on the lateral and longitudinal center lines. Following this, a 3/32" diameter hole is drilled through from each end of the plate, and these holes are then counterbored to a depth of 13/36". The pin-jack holes are then drilled in the side of the plate, using a #56 drill, and counterbored with a 3/32" drill to a depth of 5/32". This hole is tapped for the 0-80 thread on the end of the pin jack. Bracket-mounting holes are drilled and countersunk at each end of the plate for 0-80 flat head screws. The side of the plate is cut out and beveled as shown in *Fig. 2a.* This cutout is necessary when using the calibrator on RCA recording heads.

The capacitor plates are made by cutting off the heads of 0-80 machine screws, securing them in polystyrene rods which are fastened to small dural

*Audio Engineering, May, 1947

knobs as shown in *Fig. 2b*. It is recommended that the insulating rods be cut from ¼″ stock, drilling and tapping both ends before turning the material to 3/32″ diameter and cutting to the required length. The circumference of the dural knob is divided into four equal parts and stamped with small steel numbering dies in the following order, 0, 3, 6 and 9. One-quarter turn of the capacitor knob is equivalent to approximately 0.003″ linear movement of the capacitor-plate.

Before assembling the capacitor plates and rods, place a drop of polystyrene cement in each end of the insulating rod. Two collets, *Fig. 2c,* one for each of the capacitor-plate screws, are held in place by pin jacks, *Fig. 2d,* providing the electrical connection to the capacitor plates. Plugs for the pin jacks, *Fig. 2e,* are made from 1/16″ diameter brass rod. A #60 drill hole, ⅛″ deep, is made in one end of each pin to permit soldering to the flexible steel wires from the oscillator-discriminator unit. Small brackets, *Fig. 2f,* are placed on each end of the bakelite plate to facilitate clamping the capacitor assembly to a recording head.

Mounting brackets are shown (*Fig. 3*) for two types of recording lathes and are used to mount the oscillator-discriminator unit on the recorder carriage. With some types of machines (Presto 6D and Allied) it will be necessary to drill and tap a 6-32 hole in the overhead carriage and make a simple "S" bracket for mounting the calibrator. Where the instrument is to be used solely on the laboratory test bench, a small fixture similar to that shown in *Fig. 4* will be suitable.

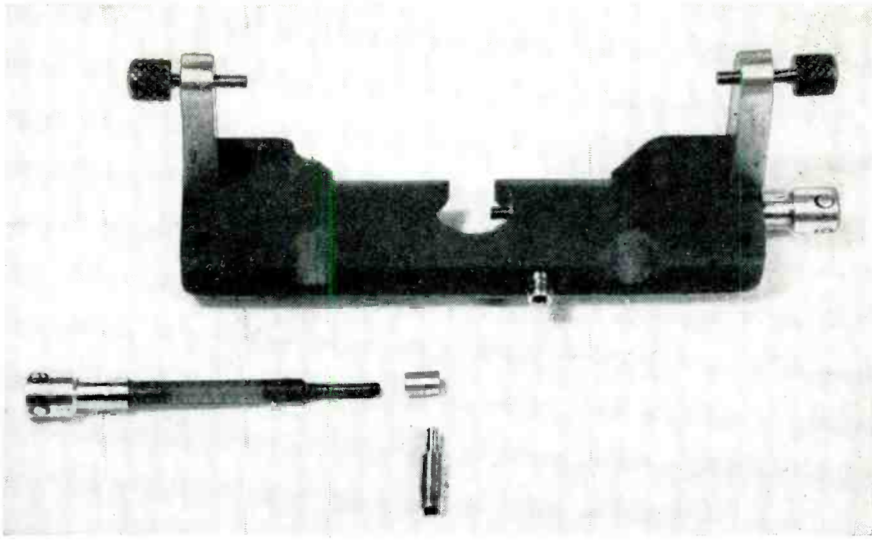


Fig. 4. Fixture for mounting unit on lab test bench.

Audio Unit

As the audio unit is a simple voltage amplifier, it will not be necessary to give data on its construction. The amplifier should be designed to give flat response within close tolerances from 30 to 15,000 cycles. The output of the audio amplifier shown in the previous article is 0 dbm, and is used in conjunction with an Altec-Lansing TI-402 analyzer which is capable of reading levels to −90 dbm. Any vacuum tube voltmeter, preferably one with a db scale and capable of reading levels to −40 dbm, will serve as an indicating device.

Since the FM calibrator is an amplitude device, the output voltage will be proportional to the amplitude of the stylus motion. This must be taken into consideration when plotting the velocity curve of a recording head and can easily be accomplished by drawing a rising 6 db per octave slope on 3 cycle, semi-log paper and plotting the calibrator readings in db above or below the 6 db per octave slope relative to an arbitrarily chosen reference which, for practical purposes, may be 1000 cycles. *Fig. 5* illustrates this graphically.

If it is desired to use the calibrator for intermodulation distortion measurements, it will be necessary to insert an equalizer in the plate circuit of the audio amplifier which will give the conjugate of the constant velocity curve of the recorder head, for one of the intermodulation test frequencies will usually lie within that range. The higher intermodulation frequency, when using a 4:1 signal ratio into the recording amplifier for testing, would be approximately 24 db, instead of 12 db, below the lower frequency at the output terminals of the calibrator's audio amplifier. Thus a seemingly high percentage of distortion would be obtained if an equalizer were not used for this type of measurement.

An improvement in the calibrator power supply, shown in Part 1, has been made recently. The oscillator-discriminator filament is now operated from a d-c supply provided by a small selenium-cell rectifier unit.

The concluding article in this series will consider applications of the FM calibrator and a discussion of some of the results obtained.

[*To be concluded*]

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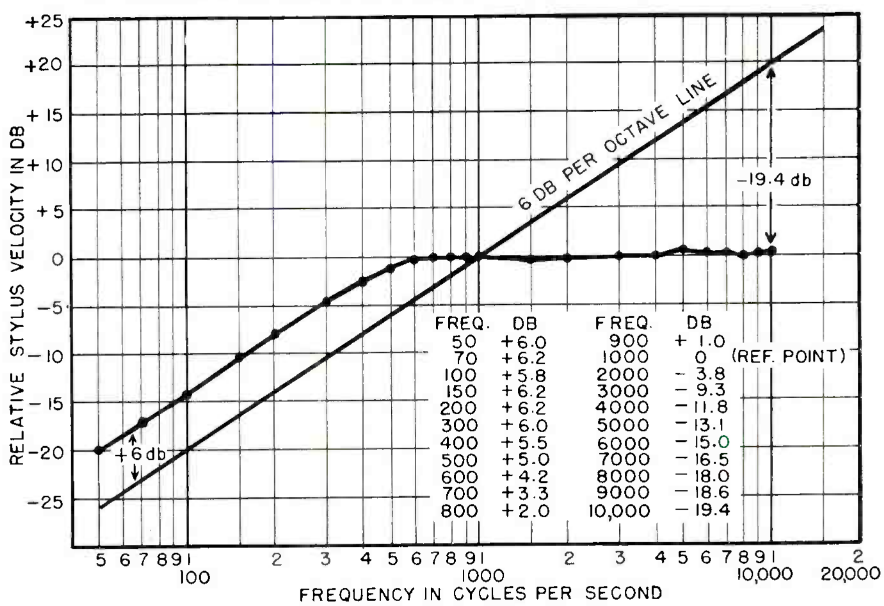
[1]H. E. Roys, "Experience with an FM Calibrator for Disc Recording Heads." *J. Soc. Mot. Pic. Eng.,* 44, 6 (June, 1945), p. 461.
[2]A. Badmaieff, "Push-Pull Frequency Modulated Circuit and Its Application to Vibratory Systems." *J. Soc. Mot. Pic. Eng., 46, 1* (Jan., 1946), p. 37.

Fig. 5. Because the output voltage of the FM calibrator is proportional to the amplitude of the stylus motion, it is necessary to plot the velocity curve of a recording head by using for reference a line drawn with a slope of 6 db per octave, as illustrated.



Perfect vs. Pleasing Reproduction*

J. MOIR

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IT IS COMMON EXPERIENCE that the majority of the ordinary members of the public prefer to operate the radio receiver with the tone control in the position that minimizes the "top" response. Discussion of this practice has probably occupied more time at "Sound Quality" lectures than any other single problem, and it may be said with perfect safety that the generally accepted explanation is that the public does not like the amplitude distortion introduced by the average radio receiver, preferring the lesser evil of "top cut." There can be no doubt that the effects of amplitude distortion, surface noise, sideband splash, etc., are minimized by "top cut," but a feeling that other reasons, possibly more fundamental, may lie behind such a universal practice led the writer to give further considerations to other possibilities.

To put the problem into its proper perspective we must have a clear understanding of our target! Is it the engineers' job to produce "pleasing reproduction" or *a perfect reproduction* of the studio performance? Purists will probably consider the implied suggestion that "pleasing reproduction" and "a perfect reproduction" are not the same thing, as rank heresy, but the writer cannot see any *a priori* reason for agreeing with this view. To do so would imply that the present range of musical instruments had reached perfection. Irrespective of the engineers' decision on this point, the public, knowing little of the fundamentals involved, will choose to have the "most pleasing reproduction" whenever the engineer places the choice in their hands. The public, having given the answer to these problems in no uncertain voice, and as it is certainly not the answer an engineer would expect, we must make an honest endeavor to decide whether the particular choice has been made because of defects in the engineering or because of some more fundamental factor.

It cannot be disputed that amplitude distortion leads to a tendency to restrict the frequency range if that facility is available. This is perfectly reasonable, for a reproducer with a range extending to 10,000 cycles per second will reproduce up to the 20th harmonic of a 500 cycle note, while a reproducer having a range up to 3,000 cycles per second can only deal with the 6th harmonic of the same note. The chance of dissonance between two notes increases with increase in their harmonic content, owing to beats between harmonics. Use of the "top cut" control will therefore decrease the unpleasant effects of amplitude distortion, and it is to be expected that its use would be more effective when a combination of instruments is being reproduced because of small differences in the tuning of individual instruments producing beats between the higher harmonics.

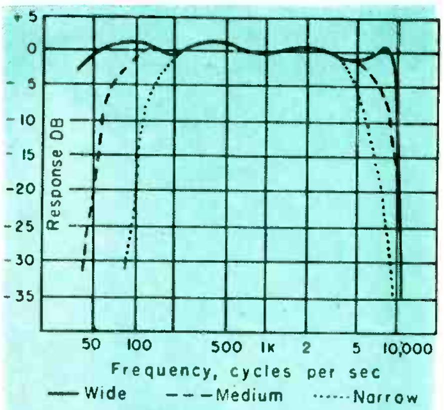


Fig. 1. Frequency characteristics used for quality judgment.

In considering the whole problem it is necessary first of all to decide what amount of distortion is just recognizable. This problem has been the target of several investigations, and to indicate the order of the problem, the results obtained by Massa (in 1933)[1] are presented in *Table 1*. The experimental method was obvious and straightforward. A high quality reproducer system was used, arrangements being made to introduce a known amount of second or third harmonic distortion without alteration of other factors. In one series of tests, listeners were asked to judge the onset of distortion without being given the opportunity of comparing the distorted with an undistorted reproduction, while in the second series of tests, a direct comparison between distorted and undistorted reproduction was possible. The tests were repeated with reproducer frequency ranges of 14 kc, 8 kc and 5 kc, and it will be observed from *Table 1* that under the stringent test of direct comparison, the minimum discernible distortion varied between 3 per cent for the 14-kc wide system and 12 per cent for the 5-kc wide system. More extensive tests by Braunmühl and Weber (in 1937)[2] gave similar results, giving support to the generally accepted figure of 5 per cent as the desirable upper limit for a high quality reproducer system.

Bearing this figure in mind we will adroitly (but with considerable misgiving) side step the question as to the relative importance of harmonic distortion and intermodulation distortion and consider the results of a more recent series of tests in which a reproducer of the highest caliber was used to determine the frequency range which would produce the *most pleasing reproduction.*

These tests[3] were much more extensive than those briefly discussed above in that separate comparisons were made on a wide variety of program material, with a series of frequency range restriction filters giving the frequency characteristics shown in *Fig. 1*. These were chosen as being representative of a system of the highest caliber (3 db down at 30 and 9,000 cycles), an average quality radio receiver (3 db down at 160 and 3,500 cycles) and a receiver midway between these limits (3 db down at 70 and 6,500 cycles).

Particular care was taken to eliminate the possible effects of such secondary factors as order of presentation, appearance of the loudspeaker, effect of signal lights, etc. High quality disc recordings have the advantage of consistency and repeatability and as a separate test indicated that the results were not appreciably modified by their use, discs were employed for the majority of the tests.

The audience chosen (about 500 people) were representative of a wide variety of possible tastes, the majority being normal members of the public. It might be expected that a specialized group of listeners, such as those owning frequency modulation receivers, would display a more elevated taste than the ordinary man-in-the-street; who might reasonably be expected to be accustomed to something worse than the narrow range reproducer. In America high quality reproduction is one of the main advantages claimed for frequency modulation technique and steps are taken to make certain that the receivers meet this claim. The even more specialized

*Reprinted from *Electronic Engineering*, Jan., 1947.

TABLE I
AUDIBLE DISTORTION LIMITS

Cut-off Frequency	Comparison with % 2nd	Undistorted % 3rd	No comparison % 2nd	% 3rd
14 Kc	5	3	10	5
8 Kc	5	5	10	7
5	12	10	17	10

group of listeners composed of professional musicians accustomed to hearing music, etc. "in the natural," might be expected to exhibit an even more refined taste. Separate tests were therefore taken on a group of people owning frequency modulation receivers and on a group of musicians.

A wide variety of program material was made available, separate tests being taken on each type, resulting in the array of results summarized in *Table 2*.

To say the least, the results are surprising. When wide range is compared directly with the medium range, the latter scores in eight out of the eleven comparisons, wide range failing to make a decisive score at any point. On the tests where wide range might be expected to score (musicians and frequency modulation listeners) it is in fact most decisively rejected, the musicians being particularly single-minded in their rejection.

On camparing wide range with narrow range the result is found to be almost as emphatic, narrow range scoring in nine out of the eleven comparisons.

The results of the comparison between medium range and narrow range do a little to restore the engineer's self-confidence, in that medium range scores in seven out of the eleven tests with one match drawn and a large percentage of the audience having no preference.

The results cannot fail to surprise the quality enthusiast, particularly as it should be noted that the usual accompaniment to "high fidelity," surface noise, monkey chatter, whistles, etc., were absent.

To the best of the writer's knowledge no comparable tests have been taken in England, but some doubts about the results discussed led to a review of some of the data presented in a paper[4] describing the equipment used by the Bell Laboratories in demonstrating the wide range stereophonic reproducer system in Washington and Philadelphia in 1933. In this instance a quality comparison was not the target, the aim being to give a demonstration of a three-channel stereophonic transmission system. As part of this system the whole of the lines amplifiers, modulators, etc., were equalized to give a frequency characteristic flat to within 1 db over the whole range of 40 cycles to 20,000 cycles, with total measured distortions of about 1 per cent and a signal/noise ratio of greater than 70 db.

The frequency characteristic of the speaker system is given in *Fig. 2* and it is noted that measured in a large room (*i.e.*, the condition of use) there is a relative loss of 17 db at 10 kc. Part of this may be offset by a rise of about 7 db in the same part of the frequency range of the microphone used, but it is particularly interesting to note that in the tone control networks provided, arrangements are made to increase the L.F. response by as much as 4 db at 40 cycles per second, but there is no provision for any similar increase in "top" response. Instead the networks are arranged to insert an H.F. cut up to approximately 15 db at 10 kc. (*Fig. 3*). Presumably this H.F. loss was only provided after careful testing had shown it to be advantageous.

In another field there is further evidence of the advantages of operating a sound reproducer with a drooping frequency characteristic at the upper end. *Figure 4* indicates the over-all frequency characteristic recommended by the American Academy of Motion Picture Arts and Science[5,5a] after careful comparison tests in several theatres. Our own experience confirms this choice.

In the space available for an article of this nature, it is not possible to quote further evidence in favor, so that we will conclude this section with the assumption that after every effort has been made to reduce distortion, noise, etc., to a minimum, the human ear still prefers the frequency range to be restricted at the upper end of the scale.

A critical review of the evidence quoted in favor of this viewpoint will immediately raise some points on which the evidence is not so conclusive as is really desirable. In the test results quoted *(Table 2)* there were no distortion or frequency characteristics taken on the specific set of speakers used for the test, though these we stated to be units of the highest quality and a group of expert listeners agreed on the excellence of their performance. These points were, however, taken care of in the Bell Laboratory series of tests, a figure in the region of 1 per cent being quoted. One might raise the objection that even this figure is too high and that it is thus incorrect to describe this system as distortionless, the restriction in frequency range being required to deal with this residual distortion. However, after giving consideration to the data quoted in the first section, it would appear that this objection is without weight. In fact after making a close study of the human hearing mechanism the writer feels surprised that 5 per cent distortion

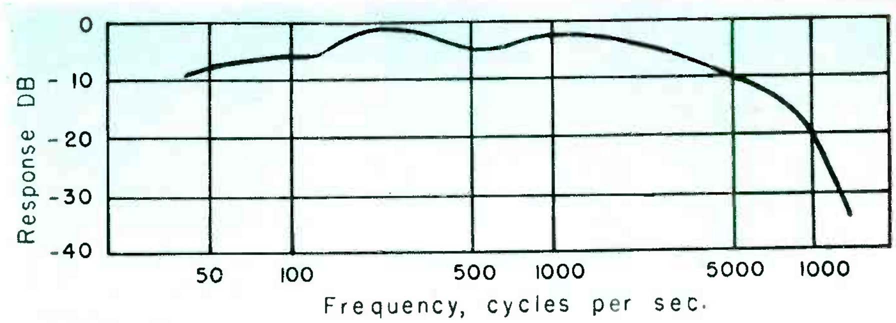
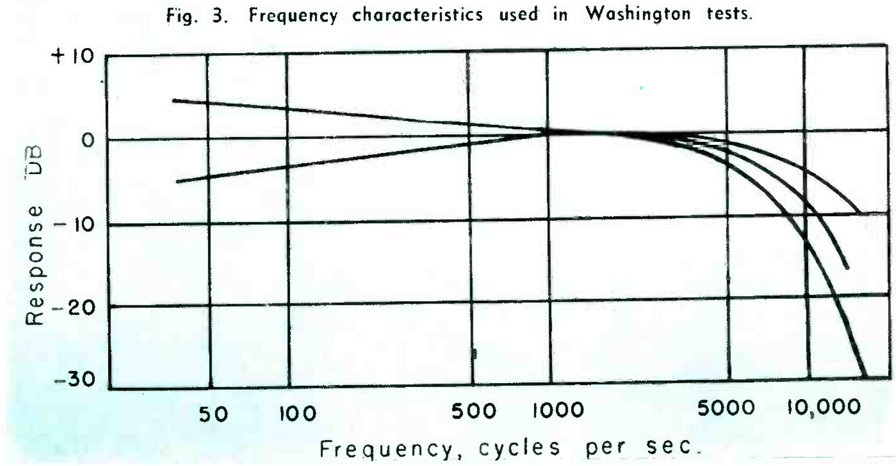


Fig. 2. Frequency characteristic of loudspeakers used in Washington tests.

Fig. 3. Frequency characteristics used in Washington tests.



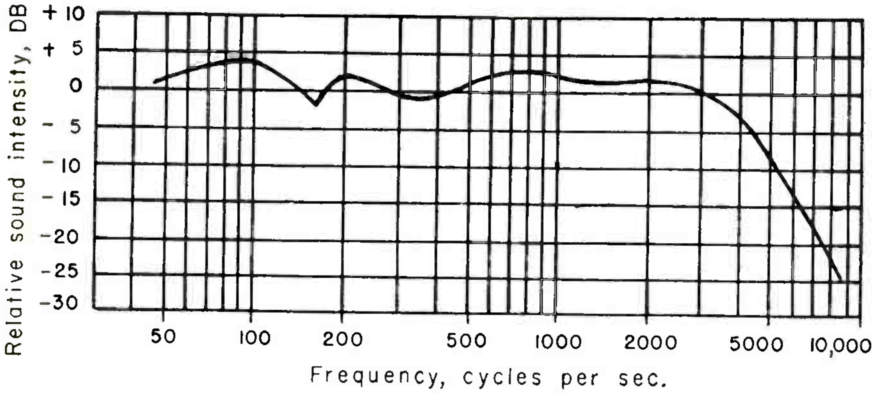


Fig. 4. Over-all frequency characteristic (adjusted for best aural results).

is detectable, as an addition to the really serious distortion occurring within the ear. In this respect it is worth noting that the tone quality (harmonic content) of almost every instrument varies enormously with loudness in normal playing and yet little impression of unpleasant distortion is given.

If our electro-acoustic equipment is really distortionless (*i.e.*, if 1 or 2 per cent is of little consequence) one is immediately led to consider whether the same preference would be shown for restricted top response if electro-acoustic equipment was not involved as a link between instrument and listener.

To the best of the writer's knowledge this point has not been checked, but an interest in the mechanics of musical instruments led to a review of some of the information that is relevant.

Violins of the old Italian masters (circa 1700), such as Amati, Guarnerius, Stradivarius, etc., have a reputation amongst musicians that is unrivaled by that of any later product, and while a lot of this may be pure "sales talk" it is apparent that a few really top rank artists can distinguish without hesitation a violin of this period from the very best copied by later craftsmen. Now it would be ridiculous to assume that the frequency response of a violin was the only factor that determined tone quality, but there is no doubt that it is a very important factor. This question of violin tone quality has been the subject of a very thorough (but as yet uncompleted) investigation by a group at Harvard University. Many lines of thought have been followed, but as our present interest is in "frequency characteristics" we will confine discussion to that factor lone.

Sound radiation from an instrument of the violin type takes place mainly from the body, the bowed strings serving only as a method of excitation. Thus, if the strings are driven at different frequencies by any convenient means the sound radiation from the body can be measured in just the manner used in taking loudspeaker response curves. This procedure has been followed for a group of the most famous models by Stradivarius, certain models by other famous masters, a selection of copies by current master craftsmen, and other models where the manufacturer had obviously kept both eyes firmly fixed on a price target. In reaching decisions on tone quality, the Harvard group had the invaluable guidance and co-operation of such eminent artists as Jascha Heifetz, Sacha Jacobson, etc.

Table 3 presents the result in tabular form for easy comparison, but a few words about the method of expressing the results are necessary before proceeding to the Table. The frequency range covered has been divided into frequency bands of roughly an octave and the figure given is the response (intensity in arbitrary units) averaged between the frequency limits quoted at the top of the column, and in order to make comparison easier the outputs have all been equalized in band 3, appropriate correction being applied to all other readings. The frequency characteristic is thus indicated directly without any further allowance having to be made for sound output differences between instruments.

Line 1 is the average performance for a group of seven of the most famous instruments by Stradivarius, instruments about which the musical world is uniformly enthusiastic.

Line 2 is the characteristic of a Guarnerius model, chosen by Heifetz foi his own use.

Lines 3 and 4 are similar data on low priced models, which the critics agreed as being thoroughly bad.

The results are illuminating, good violins are characterized by a response which falls off fairly rapidly above 4,000 cycles per second, while the bad models have a response which is well maintained up to, and beyond, 6,300 cycles. At a low frequency end the good models are again characterized by a well maintained output, while the response of the bad models fall away in the same manner as a poor amplifier.

In a good instrument the lower tone will thus be rich and full with prominent partials (or harmonics), while the upper notes will tend towards pure tones, or at least only the lower partials will be well represented. This is particu-

TABLE II
FREQUENCY RANGE PREFERENCES

Range	Classical Music	Male Speech	Piano Music	Light Music	Female Speech	Mixed Speech	Musicians Classical Music	Male Speech	Freq. Mod. Classical Music	Listeners Male Speech	Live talent Light Music
Wide	12	21	28	39	30	15	7	25	16	14	11
Medium	67	55	24	22	26	64	83	48	61	63	70
No preference . .	21	24	48	39	44	21	10	27	23	23	19
Wide	15	24	24	33	34	23	5	48	28	36	26
Narrow	58	48	30	34	29	45	73	40	59	48	40
No preference . .	27	28	46	33	37	32	22	12	13	16	34
Medium	19	52	25	33	46	34	20	62	31	55	21
Narrow	38	25	20	26	33	34	28	10	28	31	26
No preference . .	43	23	55	41	21	32	52	28	41	14	53

Wide Range is 3 db down at 30 cycles and 9000 cycles.
Medium Range is 3 db down at 70 cycles and 6500 cycles.
Narrow range is 3 db down at 160 cycles and 3500 cycles.
Figures are per cent of votes cast.

larly interesting as it is exactly the result that is obtained from a good piano.

If we assume the correctness of this thesis, it should be possible to trace similar tendencies in the trend of development of other orchestral instruments, although as musical development is an art and not a science it is to be expected that the time scale will be measured in hundreds rather than in tens of years. Nevertheless, the results can be just as positive as those obtained by any scientific process.

Tracing the development of any orchestral instrument is a rather difficult proposition, as instruments may, and do, become obsolete for reasons other than tone quality. Restricted range, difficulty of fingering or blowing, size or even mere fashion may end the popularity of an instrument.

In spite of this, the modern oboe is believed to be a fair example of instrument development. Originating from the 12th century Shawm, a conically bored instrument with a double reed, the instrument is described by Geiringer as "having all the power and astringent vigor demanded by the age," or as having "a shrill bleating tone." By another authority[8] as "an excruciating noise much more penetrating than a trumpet." By the 17th century it had become advantageous to modify the reed and mouthpiece, making (to quote Geiringer[7] again) "It is softer and more delicate," a certain indication that the top response had been modified by the removal of some of the upper partials.

By the 20th century the instrument had become the modern oboe with a characteristic, common to most of the

[Continued on page 41]

TABLE III
FREQUENCY RESPONSE OF VARIOUS VIOLINS

Frequency Range	196 cycles to 349 cycles	349 cycles to 784 cycles	784 cycles to 1568 cycles	1568 cycles to 3136 cycles	3136 cycles to 4186 cycles	4186 cycles to 6272 cycles
Average Stradivarius	14.1	23.7	25.7	25.9	23.6	10.9
Guarnieri's Heifetz choice ..	15.3	27.2	25.7	28.7	21.4	10.85
Bad Violin No. 1 ..	12.12	16.1	25.7	20.4	19.7	15.35
Bad Violin No. 2 ..	3.2	25.1	25.7	51.9	20.9	20.1

Figures are Intensities in arbitrary units.

Report on Dr. Harry F. Olson's Listener Preference Tests*

Estimates indicate that there are about sixty million radio sets and combinations in use today, with a frequency response generally limited to 5 kc. or less.

Furthermore, subjective tests of frequency range preference of reproduced sound have been made by various organizations from time to time. These tests appear to indicate that the average listener prefers a restricted frequency range in monaural reproduced speech and music. The question arose as to why such preference for restricted range was manifest, and whether it was fundamental or due only to equipment faults. Dr. Olson began this study before the war; interrupted during the conflict, it was recently completed.

Three reasons may be advanced for user preference for restricted range:

1. The user, by long listening to sets of such restricted range has become conditioned to preferring it.
2. Musical instruments have excessive high-frequency response.
3. The distortions produced in the reproducing system may be highly distasteful to the public, and a restricted range will reduce their unpalatability.

*Based on a paper presented by Dr. Harry F. Olson of RCA before the Acoustical Society of America meeting in New York City May 8-10, 1947.

Examples of such distortion are:

A. Frequency—amplitude
B. Input—output non-linearity
C. Phase
D. Single channel system (monaural)
E. Transient
F. Noise
G. Spatial distribution
H. Acoustical conditions in two spaces
J. Limited dynamic range

These sources of distortion could be eliminated by a direct air-to-air test with an orchestra in the same room as the audience, using an adjustable low-pass acoustical filter in the intervening space, and that is what was done. The pure frequency response reaction could then be tested.

The room used was comparable to a typical living room in size and reverberation characteristics, being twenty-four feet long, twenty feet wide, and nine feet high. The six-piece orchestra used for the tests was placed in one corner, and the listeners diagonally opposite. In front of the orchestra was placed the acoustical filter. The filter was hidden from the audience by an acoustically transparent cloth screen, and filter position was indicated by a pair of lamps, marked *A* and *B*, arranged one over the other.

Room reverberation time curve was shown. Reverberation time was about .8 second up to 1000 cycles, then decreased slowly to .7 second at 3000 cycles. The acoustically transparent screen had practically no absorption with respect to frequency below 10 kc and about 2 db at 15 kc.

The low-pass acoustical filter consisted of three rows of plates, suitably perforated. The six-foot "tone control lever" swung the plates either to fill the open spaces in the partition, or at right angles to the partition to leave an unrestricted sound path. Filter attenuation curve was given; it showed uniform transmission to over 4 kc, rapidly increasing attenuation above 5 kc, reaching 40 db at 10 kc. In electrical terminology, curve was such as would be produced by $Z_1/Z_2 = 4$ at 4 kc.

Orchestra consisted of the following instruments: piano, clarinet, trumpet, violin, double bass, drums and traps; players all had professional experience. Most pleasing acoustic peak intensity level was between 70 and 80 db; the orchestra had to do considerable practice to get to this low level in so small a room. Several well-known musical directors checked the resulting tone

[Continued on page 43]

Simple RC Filters for Phonograph Amplifiers

The circuits described in this article are designed to enable excellent reproduction from the new magnetic type pick-ups.

GEORGE L. ROGERS*

Many radio men who are interested in playback of recordings, for themselves or customers, have encountered trouble with "fuzzy" high-frequency reproduction. Some notice it soon after adding a wide-range speaker to the system. The trouble referred to is one which persists after all units of the system seem to be all right. It shows worst on poor pressings, and even on good pressings after they become a little worn in the loud passages. In both cases the distortion is most apparent in loud passages near the center of the record. Listening tests made with the best available reproducer show that only the best commercial releases have any usable overtones as high as 10 kc, and on many the surface noise and distortion above 6 kc mask any usable overtones. Some are playable only when the system is cut off above 4 kc. Of course, some of the distortion is due to the inherent characteristics of disc recording and is always present, regardless of the quality of the recording and playback.

Frequency Response

However good or bad the system may be, the response of the playback should be cut off sharply at some frequency, for this gives the best reproduction for a given amount of distortion and surface noise. The same applies to any system, whether film, wire, or even live talent through a microphone-speaker

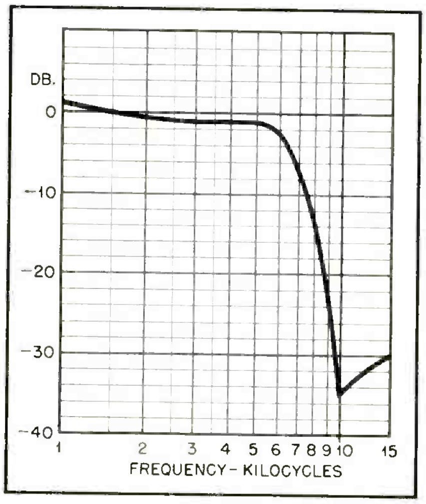


Fig. 2. Reponse curve of amplifier and filter shown in Fig. 1.

system. If the high-frequency end drops off relatively gradually, as is the case with any amplifier without a filter, the distortion and surface noise on average pressings cannot be sufficiently reduced with simple RC tone controls without impairing the medium high frequencies between 3 and 6 kc.

All this is well known to those who have dynamic and variable reluctance pick-ups and filters at their disposal. But these items are not usually available to the average individual, and he usually plays records at high level to make the distortion less noticeable. Even with the proper filter, records sound better at high levels, but distortion is reduced at all levels, the improvement becoming more noticeable as the level is reduced.

The pick-up situation has finally improved with the recent appearance of new units which give smoother high-frequency reproduction than those previously available in the low-price range.

But the cost of filters and filter components is still high and they are not always easy to obtain. Some inexpensive pick-ups use mechanical cut-off to avoid the use of an electrical filter, but this is usually unsatisfactory for one or more reasons. The mechanical filter may lead to excessive needle-talk because of reduced needle compliance, or harmonics generated in the pick-up element may appear in the output even though they are above the mechanical cut-off fre-

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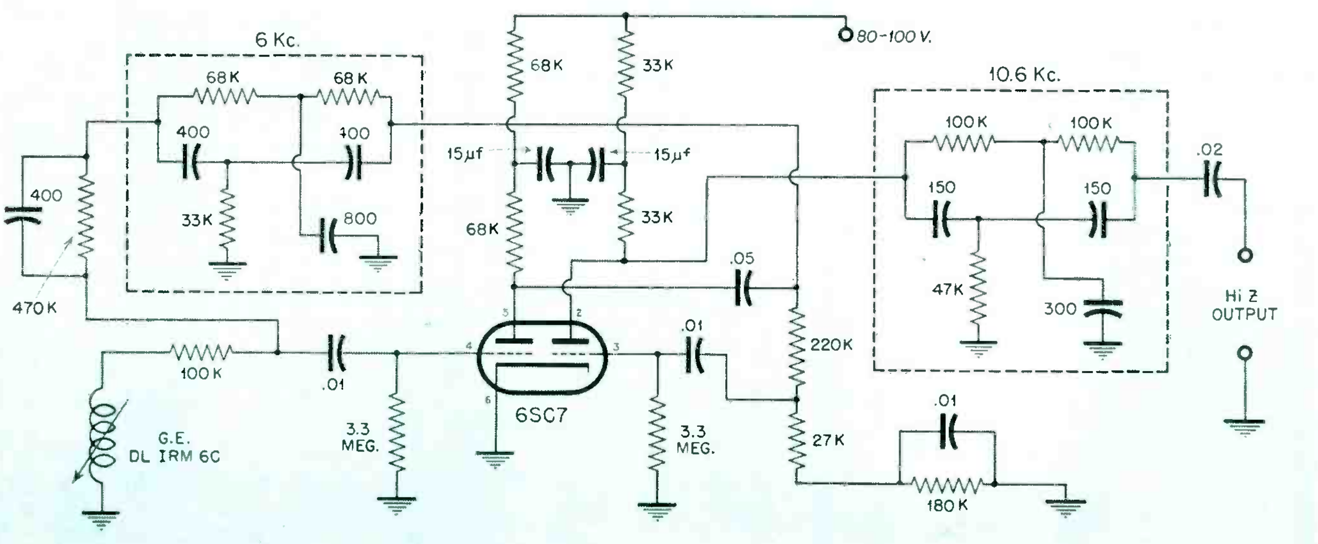
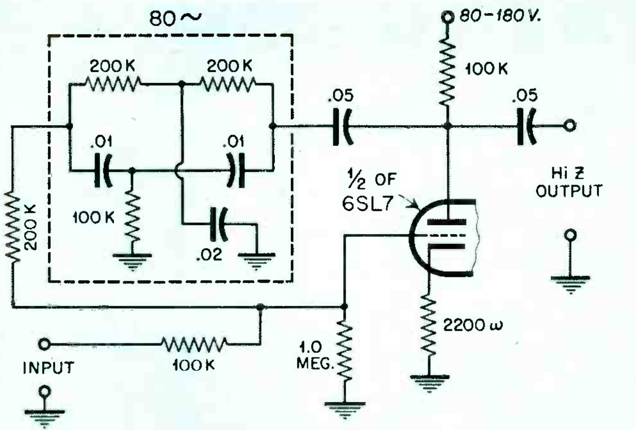
Fig. 1. Schematic of preamplifier and filter circuit especially suited for magnetic pick-ups.





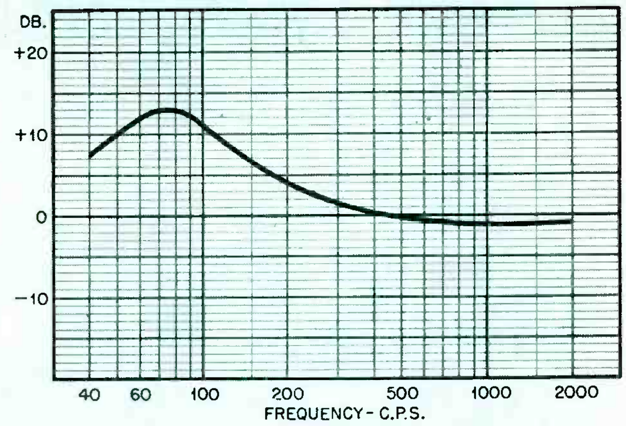


Fig. 3. Circuit to provide bass boost and (right), Fig. 4, its response curve.

quency. The best results are usually obtained with a wide-range pick-up and an electrical filter to give the cut-off.

One object of this article is to present an inexpensive RC low-pass filter to use with low output variable reluctance pick-ups. The circuit (*Fig. 1*) is similar to that of preamps recommended for use with this type pick-up, with the addition of two conventional parallel-T networks to provide a low-pass filter action. The 6-kc network in the feedback loop gives a 6-kc peak and the 10-kc network gives cut-off at approximately 10 kc. The resultant response (*Fig. 2*) shows satisfactory output to 6 kc and fairly sharp cut-off above 7 kc. This cut-off is gradual compared to properly designed LC filters, but gives substantial improvement in reproduction compared to most RC filters of equal cost and simplicity. Naturally feedback reduces the gain, but the output is still sufficient to satisfy most amplifiers. The high impedance of the output of the filter is a convenience in most cases, but the output should be coupled to the grid of the following stage with short leads to avoid the shunt capacity of long shielded leads. If filament hum is encountered it can usually be cured by grounding the center tap of the filament circuit either at the transformer or through resistors.

The parallel-T feedback loop also is a convenient way to obtain bass equalization of boost without using a choke. The circuit (*Fig. 3*) uses a network tuned to 80 cps. The amount of the rise may be controlled by varying the amount of feedback, and the frequency of the peak may be shifted by changing values in the network. Response is shown in *Fig. 4*. This type circuit is preferable to the average RC boost circuit because the 400-cycle region is not lifted objectionably and the response falls off below the peak, giving less turntable rumble for a given boost at the peak. Also there is less "hangover" than with some tuned LC circuits.

Another use of the parallel-T is to cause a dip in the response of the amplifier. In this case the network is used without any tube and the amount of dip is controlled by resistance between the two terminals above ground. This is useful in instances where the speaker or room, etc., have objectionable resonances.

Many variations in the above circuits are possible to vary the frequencies of cut-off or maximum transmission. The network and application in feedback am-

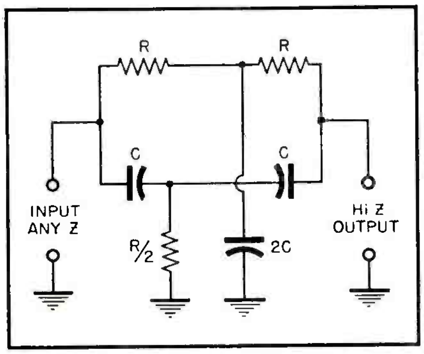


Fig. 5. Fundamental filter circuit.

plifiers have been analyzed in various textbooks and technical publications, but the fundamental formula for determining the null frequency is repeated here for convenience. In *Fig. 5* the null frequency $f = \frac{1}{2\pi RC}$ where R is in ohms and C in farads. A more convenient form is $f = \frac{159{,}000}{RC}$ where f is in cps, R in ohms, and C in microfarads. The null occurs at this point and transmission returns on each side of this point.

No matter how the constants of *Fig. 1* are varied, it cannot be expected to duplicate a good LC low-pass filter. The main use of these filters is to improve results in medium-priced audio systems such as PA and home phonographs. They have the advantage of low hum pick-up, price, and space requirements, and are useful when the results are good enough to be consistent with the quality of the associated system.

Troubles With Records

Since the above has been non-mathematical and the merit of results largely decided upon by listening tests, some of the inherent distortions in records might be discussed from the same viewpoint.

There is a widespread belief that if record processing and playback were perfected, the results would be nearly life-like. This may be true, but some of the weaknesses in the basic principles of disc recording will make this very difficult to achieve.

If commercial releases are to be improved, several steps are necessary, and widely known and discussed. Probably the most universal offender is the surface. Vinylite pressings have been sold and some show the results expected in reduction of surface noise. Some were disappointing because of impurities in the vinylite or bad pressings. Those with impurities were noisier than the average shellac. On the whole, the average vinylite has not realized the possibilities. However, when the surface is good, vinylites are superior to competitive shellacs.

Frequency range is an old bugaboo, and the differences in range and recording curves of different manufacturers have been well covered elsewhere. New cutters promise improvement in the manufacturing end. Some new cutters for 78 rpm lateral work employ electrical feedback and multiple windings in the cutter, and lead the way to recordings with the turnover point and the velocity at high frequencies controlled electrically rather than by mechanical damping.

Reverberation is a factor of which

[*Continued on page* 47]

Opportunities In Ultrasonics

S. YOUNG WHITE

Ultrasonics has many important industrial applications which are now being developed. In this article the author discusses some of its potentialities and how they may be used.

THE ENGINEER working in the audio spectrum finds a queer world indeed when he considers entering ultrasonics as a business. The field seems so specialized and the uses so limited that it is not very attractive at first sight. However, there seems to be a considerable amount of activity in the art, and the audio man is certainly the one to enter the field when it is ready for commercialization.

In this article we shall attempt to answer the obvious questions about the new art—how it differs from audio, the range of values that seem useful, the basis of its potential usefulness, and how soon we can expect to see it become a business of some commercial importance.

First, the name "ultrasonics." Ultra means beyond, of course, but for many years it was known as supersonics, the waves were so high in frequency that they were beyond audibility, at least, to the human ear. A few years ago the aircraft boys began to speak of speed of flight as more than the velocity of a sound wave, and called it supersonic flight. The publicity they obtained for the term was so great that they usurped the term. The Germans had always called it ultra-schall, or ultrasonics, so many of us have followed suit.

The lower limit of ultrasonics has never been standardized. Very few people hear frequencies above 15 kc, so for most people ultrasonics begins at this point. Cut-off at 18 kc is fairly common, however, and some women can hear to about 24 kc. So frequencies above 24 kc can safely be called ultrasonic. As mentioned in last month's article, however, at very high levels the cut-off frequency of any ear seems to be lower than for a normal signal level.

A sound wave, of course, is a compression wave followed by a rarefaction wave traveling longitudinally in an elastic medium. It is strictly mechanical motion and will not travel through a vacuum. An ultrasonic wave is the same, except the frequency is above 20 kc, let us say, and the waves are correspondingly short.

The upper limit of ultrasonics has yet to be reached. It is quite easy to work to 7 megacycles, especially in water, and 50 megacycles have been reached. The upper limit in a particular medium is probably where the wavelength becomes as short as a molecule—many hundreds of megacycles.

The first point of difference with the audio art is that the audio engineer is the slave of the human ear. He must maximize pleasant sounds and minimize unpleasant ones, and no matter how many instruments he has as a guide, the final arbiter is the ear of the listener. In ultrasonics we cannot be heard in any case, so all work is by instrument. Also, we have no weighting of the spectrum to fit the ear characteristic. We have the incidental advantage that we can use high levels without bothering anyone, although below 35 kc or so the canary birds are annoyed, and below 30 kc dogs are affected.

Applications

With the limitation of inaudibility, what uses are left for sound wave? There are two main fields, measurement and processing.

The field of measurement is mainly based on the fact that at high frequencies the waves are so short that they can rather easily be directed or focused—precisely the reason we go to microwave transmission in radar. A one-kc wave in air is a foot long, and in water about five feet, so a parabolic reflector, or a diaphragm large enough to send out a few degree plane wave beam would be many feet in diameter. At one megacycle, everything becomes a thousandth the size, which is a vital improvement.

Processing is a field of unlimited possibilities. By applying ultrasonic power to a material the characteristics of the material are altered in some way. The effects are all based on the tremendous acceleration forces that can be applied to the material in this manner.

The only field in which any money has been spent has been the wartime use of the pulsed ultrasonic waves for detecting submarines, and substantially the same apparatus for bouncing waves off the bottom of the ocean. In each case highly directional beams are used, and the time for a round trip of the pulse is translated into distance.

The historical background of the ultrasonic art is quite old. Several British patents* of 1912 show it being used both in air and under water for detecting objects ahead of a ship. These make interesting reading, as he ties in the ultrasonic art with the known audio theory very well. Langevin, with his French patent 502,913 (1918), shows a quartz crystal transducer that detected submarines passing within several miles of the coast.

Between wars the art languished, as the field was pretty well limited to the military boys, who could not obtain very large appropriations. But there was some interest among the physicists, who used it to investigate the properties of matter. There are about 1200 published papers on the art, giving us great hope of useful results. About two-thirds are on measurement, but the rest show processing effects that can be of first-class commercial importance.

Testing With Ultrasonics

Let us first consider ultrasonics for measurement work. There are two main applications here; testing material by measuring the time of travel of a pulse through it, and by measuring the absorption of a wave in a known length path through the material.

The velocity test is possibly most practical in a gas mixture. The velocity in gases met with commercially varies about 14 to 1, so it is easy to visualize measuring the percentage of each gas in a two-gas mixture by measuring the velocity of the mix. The main difficulty here is the large error introduced by the large temperature coefficients. For air this is one per cent for each 10° F. change. Either we must have an automatic corrector or we must maintain the gas at constant temperature.

In measuring liquids for velocity the range is smaller, about 2½ to 1, but the effect of temperature is often even more pronounced. However, many commercial processes already control the temperature of the mix to within 2°.

These measurements can be made with quartz or PN crystals for pulse transmission and reception, the PN working up to 100° C. Often we set up an oscillator whose frequency depends

*Richardson, Nos. 9,423 and 11,125.

on the time of transmission of the pulse, or we can compare the time with that of a standard path in parallel with the test path.

The absorption effect comes in when the liquid has small particles in it, such as paper pulp, and we wish to know what the percentage of concentration is. The larger particles, such as cornstarch, can be measured at 50 kc, but for colloids in small concentration it is necessary to go to the highest practical frequency, which is 7 mc. This will measure a tenth of one per cent of Kraft paper stock or a true colloid, such as titanium dioxide.

The photograph shows some experimental equipment necessary to measure the percentage of particles in a liquid, such as paper pulp in water. It seems there are numerous devices to measure this concentration when there is about 4% pulp, due to the change in viscosity, but they actually make the paper with only one-half per cent pulp.

Some experience gained with this setup showed us that we must have exact service conditions to make such an investigation effective. We must have a flow, the gas in the water must approximate that in practice, we must cancel out temperature and pressure effects, and handle a large number of other elements in the water, such as organic acids.

Preliminary work only can be done in a non-flow test chamber. A controlling factor is the wetness or degree of hydration, of the particle. Dry corn starch wetted and stirred for one minute has nearly a hundred times as much absorption as when thoroughly wet, which takes about four hours. So the system is only good for material which is made wet and stays that way all through the process. On the other hand, we can measure the degree of hydration if we must wet material that comes to the factory in the dry state.

This illustrates a side of ultrasonics that is holding it back from immediate use in many instances. We are dealing with the fundamentals of matters, and there are many things not known about colloids, their boundary layers and so on, so it is necessary to try each one. Some materials give no trouble at all, but at this stage of human knowledge we do not know why.

Processing

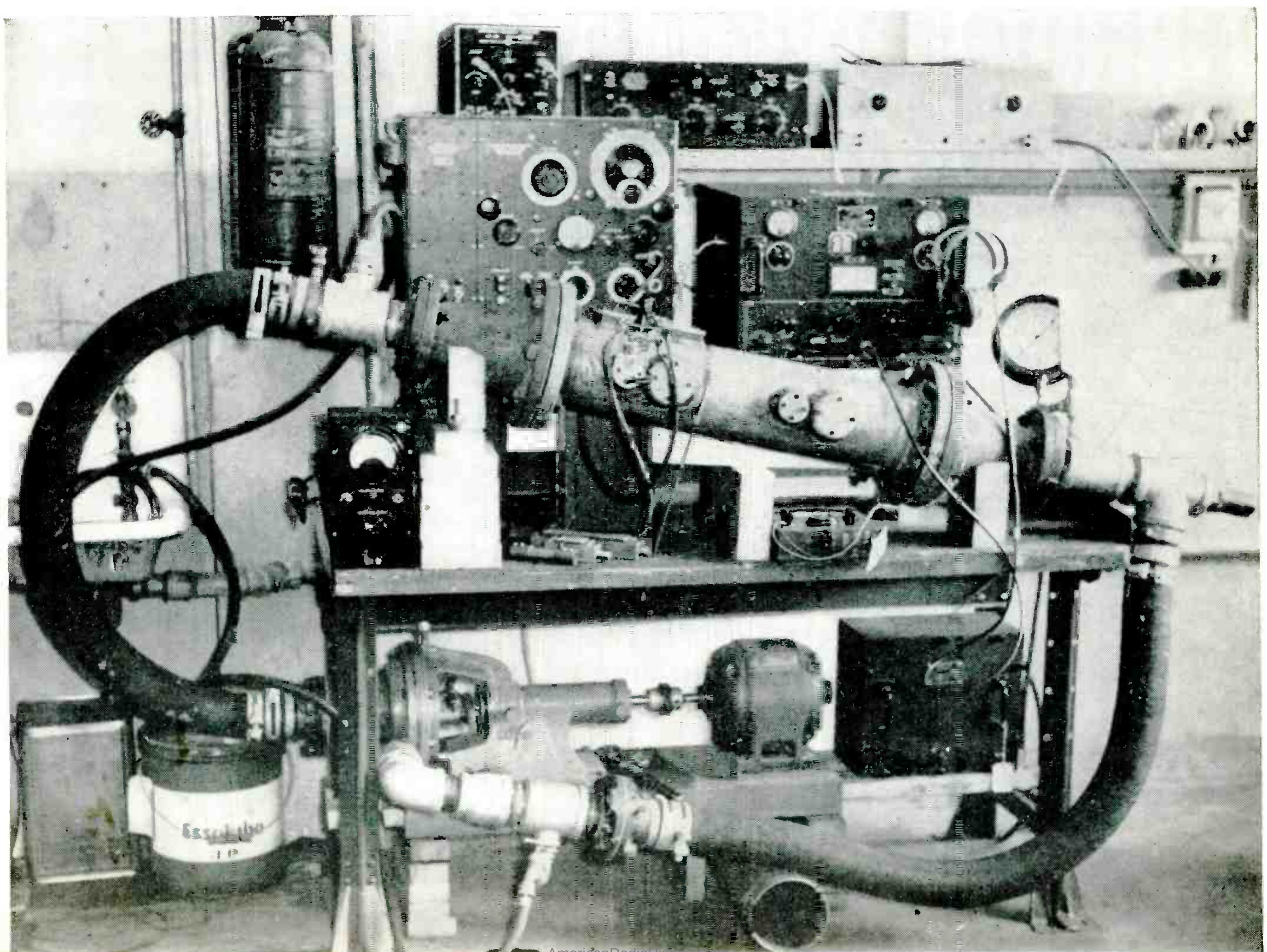
The second and potentially unlimited use of ultrasonics is for processing. The effects noted in the laboratory are endless, and many of them highly desirable commercially, from bacteriology to metallurgy. These all stem from acceleration effects, so let us see what order of effects we can hope to obtain.

The table (*page 32*) is calculated from formulae given in Bergman's "Ultrasonics," a very good work with a bibliography of about 600 papers.

To tie this table in with common experience, let us assume a ten-inch cone speaker putting out a half watt at 1 kc. This would usually require ten watts in the voice coil. The energy density would be (if the cone acted as piston, which it does not) 1 milliwatt per cm^2. This would give us a motion of just under 1 mil, and an acceleration of 2 miles a second a second, or 140 G.

Also to remind you of your high-school physics, the acceleration of the shell in a gun ten feet long is about 40 miles per $second^2$. If we take the case of a power density of 10 kw/cm^2 at a frequency of 150 kc, since the pressure goes up as the square root of the power density and we have 10,000 times the power, the increase is 100 times, giving us 13,000 pounds per square inch plus or minus in steel, and 2500 in water. For air at zero gauge it becomes the impossible value of 41 pounds per square inch, and we have only 14.7 pounds pressure in the room. Since we also increase the acceleration directly with the frequency, the increase is in round numbers 150/24, or 6 times. So our acceleration is times 100 for the increased power, and times 6 for the increased frequency, so we come up

Ultrasonic generator set up to measure percentage of particles in a liquid.



with a figure of about 60 million G for air, or 360,000 miles per second^2.

Let us clarify the difference between power and power density. Power, of course, is the total amount available for work. But power density can be thought of as power in a correctly terminated line on its way from the generator to the load, with zero reflection. If this line is terminated with a resonant tank circuit with a Q of 100, then if the impedance match is correct, 100 times the energy will circulate as wattless component. This concept is of great commercial importance, since most loads will have high Q values, and when treating tons of material a day it would take too much power to establish enormous wattage density per square cm unless there was the resonant rise to help by a very large factor.

These figures give the story as to the difficulty of applying ultrasonics to the production line. There are three main problems:

1—Difficulty of generating the power. For measurement work this is no problem, as for most cases magneto-striction or piezo crystals are a good enough answer. For generating a few kilowatts they are almost useless. The gas generator shown in the last issue is an answer, however, as it can be made in almost any power, and can run on steam, the cheapest primary source, but it is limited to gas loads.

2—Impedance match, and

3—Boundary layer problems.

The impedance match prevents the gas generator from being of much use in solids or liquids. By looking at the table, it is easy to see there is some chance of transferring energy from water to steel, as the motions, pressures and acceleration are not hopelessly different. But gas (air) is so far out of line it is hopeless.

The sonic impedance of a material or medium is the density times the sonic velocity. In British units, the value for steel is 8 million, water 0.3 million, and air only 90. So when we send a wave from one medium to another, the boundary layer reflects energy roughly proportional to the square root of the two impedances. For instance, in going from water to steel, or vice versa, 85% of the energy is reflected. So the reflection from water or steel to air is practically complete. This means we cannot kill a man at any distance with ultrasonics, as practically all the energy would be reflected from his skin.

The boundary layer problems, aside from impedance match, are very difficult to overcome. Even in measurement work, we must use a few milliwatts of energy over a small surface, and if we use the megacycle region we must accelerate the material to very high values.

These acceleration forces are so high they are beyond usual experience. They are common in explosions and similar destructive phenomena, such as collisions. But here we have a nice tame effect continuing for long periods, and any effect per cycle is continued for millions of cycles in a few seconds at least. So particles are torn apart, rubbed on each other, their electrical charges affected, heated, all to an amazing degree. And all these effects are going on at once.

If we use a diaphragm on our transducer, and immerse it in such an ordinary liquid as tap water, we find the most unusual chemical combinations take place. There are two dozen chemicals and compounds in New York City tap water, from chlorine to organic acids, so a monel diaphragm will have deposits on it which are green one day, red the next, and so on. If we do not excite the diaphragm, it remains clean.

Cavitation

Another great diaphragm problem is cavitation. The pressure in the room is 14.7 lb/in^2. We usually consider we modulate it with our sound wave up and down from this value in a symmetrical manner. But it is obvious that at about 2,000 watts/cm^2 we have a sound wave completely modulating this pressure, and beyond this we can compress the air any amount we wish, but we cannot go below zero with negative pressure. At this point cavitation begins, where the air is torn apart, leaving vacuum cavities.

A look at the table shows that in water only one watt gives us minus 25 lb/in^2, so water in a pipe at ordinary city water pressure will cavitate at slightly above one watt. But there is another effect even more pronounced. The attenuation of an ultrasonic wave in water is very slight, so in a chamber of the necessary limited size we can often set up a standing wave pattern of very high amplitude. If this becomes out of phase with the energy fed in through a diaphragm there will be a cavity formed in the water before the diaphragm, and then the out of phase motions of the diaphragm and the water molecules will collide head on at greater than sonic velocity, due to their combined velocities. This causes a hammering action on the diaphragm that has developed pressures of the order of several hundred thousand pounds per square inch, etching the diaphragm severely in a few seconds.

With about ten watts per cm cavitation will tear bacilli apart, so we have possibilities of a universal sterilizer. For food in cans we must first generate about a kw by some economical means, then put the energy inside the can without breaking loose the tin layer from the steel can. Then we must insure there are no dead spots that are of so low amplitude that cavitation does not occur.

The gas generator of the last issue can be used without boundary layer problems in some applications, as we can simply pass the gas through it and obtain accelerations of about ten million G which should seriously affect any germs in the gas. A similar effect should take place in large molecule gases, where it is known about 50,000 G will break up long hydrocarbon chains.

By passing mixed gases through the generator the speed of reaction between them should be considerably increased, as has been noted by some experimenters. It is a curious point that many gases do not obey the gas law of volume, pressure and temperature with U-S, as it takes many microseconds for some gases to heat up after the pressure is applied, so by working up around a hundred kc the gas heats up only negligibly.

After this quick look at some of the applications and problems, how does U-S stack up as a business?

The difficulties are very great. Much work must be done on instrumentation. A high-power solid or liquid type generator must be found, and the boundary layer problems licked by earnest effort

[*Continued on page* 47]

BASIC VALUES OF ULTRASONICS AT 24 KC

For a power of 1 watt per square centimeter.

Medium	Pressure *Lbs/in²*	Motion of molecules *Micro-inches*	Acceleration *G*	Acceleration *Miles/sec²*
Air	± 0.41	1830	107,000	620
Water	± 25.0	30	1,700	10.5
Steel	±130.0	6	350	2.1

To convert to other values.

Values increase as the square root of the power density; that is, at 100 watts/cm^2 all values would increase ten times.

Acceleration increases directly with the frequency, and the motion inversely as the frequency. Frequency has no affect on the pressure, however.

1 watt per cm^2 is equal to 6.5 watts per inch^2.

RECORD REVUE

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

As most engineer readers of this magazine are aware, there are major developments now under way in the low-priced pickup field which in time will have considerable effect on the design and performance of the average home phonograph. The various principles involved, most of them magnetic, are well known. From the music listener's point of view here are the important considerations in this field:

All the new-type pickups run lightweight; they all involve permanent or at least semi-permanent styli. ("Needles" is hardly the word now, when nothing is left but the sapphire point of a needle!) But aside from the markedly smaller elements involved, the factors of weight and permanency are not particularly new; lightweight crystals with built-in needles appeared before the war and the revolution of weight may be said to have been in the main accomplished.

Much more important in these new pickup cartridges are the factors of adaptability and of performance. Virtually all of the non-crystal pickups, both cheap and fancy, thanks to their very low-voltage output and their different response curves, cannot be used directly in place of the usual crystal pickup now found in 99% of existing machines. Thus for the time being, until new types of commercial machines are made with built-in preamplifier and equalizing stages, the mass market buyer is likely to steer clear—unless radio servicemen discover a bonanza in "conversion" jobs at a suitable price! The cartridges will, however, sell fast to those who are able to do their own converting or who can install a ready-made preamplifier.

But, in spite of these drawbacks and others of a technical nature, the new pickups are here to stay, because under good conditions they give remarkably improved reproduction. The most arresting improvement is that of fidelity. The new type pickups are inherently almost limitless in their response over the audible range and can give high fidelity performance as a matter of course.

The fly in the ointment here is the rest of the equipment. Most existing phonograph amplifiers are not up to the new pickups, and many a buyer is doomed to disappointment when his new pickup carefully reproduces for him all the distortion that the old one mercifully ignored! But a triple demand—the new pickups, plus FM radio reception and the higher range records now made—is bound to force some radical changes in audio amplifier design before long.

Given a proper preamplifier, equalization, amplifier and speaker, there still remain some remarkable performance differences between the newer pickups and former types. Two are startling in their effect on the listener—the difference in the amount of needle chatter or talk, and the similar difference in the amount of needle resonance that gets into the amplified signal. Many listeners are not consciously aware of it—until it is removed. The new types are almost entirely silent when passing over even the loudest passages on a record. The effect is nothing less than startling to a hardened record listener! Perhaps even more remarkable is the entire absence of the needle resonance that carries over the peaks and buzzings of needle talk into the signal voltage and hence into the speaker itself. The net effect is that the listener is scarcely aware of the actual pickup and record at all—the music holds the field to itself.

A final and somewhat unlooked-for feature of the new pickups is, strangely enough, a relative *reduction* of surface scratch. One might well expect, with an extended range pickup, to run into greatly increased scratch, especially on old records. Naturally, as compared to a low cut-off crystal, there will be an increase of scratch noise. But, taking the response into consideration, the actual scratch that appears is remarkably easy on the ears. The impossible is accomplished partly by the needle shape itself, partly by lack of response to vertical scratch components, and as far as I can see, partly by smooth riding qualities—the scratch that does come through is smoother and more even. Whatever the reasons, this is a highly welcome feature.

I am aware that numerous disadvantages have already appeared to plague those who are experimenting with one or more of these new pickups; I am sure, however, that these disadvantages

[*Continued on page* 44]

MUSICAL ACOUSTICS

BENJAMIN F. TILLSON*

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

PART I

AUDIO ENGINEERING must take full cognizance of the mathematics, physiology, and psychology involved in the appreciation of musical sounds. In no other way can the nature, need for, and demands of high-fidelity sound reproduction, transmission, and reception be given thorough consideration.

Since some audio engineers enter that field with a background of theoretical and practical music, others as physicists, and still others as electrical engineers or architects, some parts of a presentation of this subject may reasonably include information known to any specific group, and yet brevity will cause much to be left out with which each may be familiar. Perhaps an emphasis on certain phases of music theory and psychological aspects will best add to the knowledge of the majority.

The arithmetic of musical intervals is considered at such length because it seems to have been so generally neglected in the education of all of the above professions. Grove's dictionary may express the attitude: . . . "the instrument maker needs the physical knowledge, the instrumentalist is the better for a little of it, the composer can afford to ignore it. No physical conclusion is of musical value which the ear does not endorse."

Obviously, without the practical application of such studies the public could never learn to enjoy the virtues of "just intonation". The human ear needs training, and the composer is no exception. Now our melodies and harmonies are shackled by the limitations of archaic, keyboard musical instruments and the music staff symbolism designed for them, as well as the literature composed for performance upon them.

In an era when electronics offers such glorious opportunities for progress, it is anachronistic for music to remain medieval. Eight times our present number of intervals per octave are discernible to the normal ear, and they can readily be produced electronically. By electronics an infinite number of tone colors can be produced by the presence of varying proportions of the harmonic over-tones. The tonal effects of the most acceptable violins and other instruments can be reproduced without any sacrifice of the ability of the individual musician to express his artistry. The infinite harmonic possibilities should prove a challenge to lead composers out of their present-day mediocrity. But of course keyboards and twelve-tone octaves must be abandoned in such a future.

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TABLE I

Tone	Equal Temper	Just Intonation Scale I		Scale II				Scale III	
	Frequency						Keys	Frequency	Keys
C	1.0000	1:1	1.000	1:1	1.00	1.0000	C.G.F.B^b	0.98767	E^b.A^b.D^b
$B^{\#}$						1.00113	$C^{\#}$		
D^b	1.0595	27:25	1.080	16:15	1.067	1.0535	A^b.D^b.G^b.C^b		
$C^{\#}$		25:24	1.042	25:24	1.042	1.0547	D. A. E	1.06787	B. $F^{\#}$. $C^{\#}$
D	1.1225	9:8	1.125	9:8	1.125	1.1250	C. G. D. A	1.11111	F. B^b. E^b
E^b	1.1892	6:5	1.200	6:5	1.200	1.1851	B^b.E^b.A^b.D^b	1.17055	G^b. C^b
$D^{\#}$		75:64	1.172	75:64	1.172	1.1865	E. B. $F^{\#}$	1.20135	$C^{\#}$
F^b		32:25	1.280	32:25	1.280	1.2486	C^b		
E	1.2599	5:4	1.250	5:4	1.25	1.2500	C. G. F	1.26562	D. A. E. B
F	1.3348	4:3	1.333	4:3	1.333	1.3333	C. F. B^b.E^b	1.31687	A^b. D^b. G^b
$E^{\#}$		125:96	1.302	675:512	1.32	1.3348	$F^{\#}$. $C^{\#}$		
G^b	1.4142	36:25	1.440	64:45	1.422	1.4047	D^b. G^b. C^b		
$F^{\#}$		25:18	1.389	45:32	1.406	1.4063	G. D. A	1.42383	E. B. $F^{\#}$.$C^{\#}$
G	1.4983	3:2	1.500	3:2	1.50	1.5000	C. G. D. F	1.48148	B^b. E^b. A^b
A^b	1.5874	8:5	1.600	8:5	1.600	1.5802	E^b.A^b.D^b.G^b	1.56074	C^b
$G^{\#}$		25:16	1.562	25:16	1.562	1.5820	A. E. B	1.60181	$F^{\#}$. $C^{\#}$
A	1.6818	5:3	1.667	5:3	1.667	1.6667	C. F. B^b	1.68750	G. D. A. E
B^b	1.7818	9:5	1.800	16:9	1.778	1.7778	F.B^b.E^b.A^b	1.75583	D^b. G^b. C^b
$A^{\#}$		125:72	1.736	225:128	1.76	1.7798	B. $F^{\#}$. $C^{\#}$		
C^b		48:25	1.920	48:25	1.920	1.8729	G^b. C^b		
B	1.8877	15:8	1.875	15:8	1.875	1.8750	C. G. D	1.89844	A. E. B. $F^{\#}$
C'	2.0000	2:1	2.000	2:1	2.000	2.0000	C. G. F. B^b	1.97531	E^b. A^b. D^b

Sound

Sound is the sensation perceived by means of the auditory nerves; and it is normally produced by a series of compressional waves which originate from a vibrating body and are transmitted to the ears by elastic mediums, of which air is an important one. Sound vibrations radiate in straight lines from their source, and their transmission of energy is like that through a line of contacting billiard balls when an end one is struck.

Musical sound, Tone, results from a regular or periodic number of vibrations per second; and Noise results from irregularities in the period of the vibrations.

The upper audible limit of sound is reputed usually to be about 33,000 vibrations per second; but very few people, and those youthful, usually can hear 20,000; rarely do those past middle age hear a higher frequency than 15,000. Although the lowest frequency of vibrations commonly acceptable as observable by mankind as a musical sound is about 32 vibrations per second, the lowest note on the piano, "A", the seventh space below the bass staff, emits about 27; and one is led to believe that he could hear still lower. But perhaps its fundamental is not heard, and the tone is identified by the higher harmonics which are present.

The highest fundamental tones produced by musical instruments are the "D flat" of about 4,450 vibrations per second produced by the E flat band piccolo, and "C" of 4,186 vibrations poorly attempted by the piano. In music notation, that D flat occupies the tenth space above the treble staff. The range of musical instrument fundamentals is, therefore, considered seven and one-third octaves. Middle "C" of music notation is placed on the first ledger line above the bass staff and, correspondingly, on the first ledger line below the treble staff. Its frequency on the basis that the "A" above it is tuned at 440 vibrations per second is variously 261.6 for a "tempered scale" or 264 for "just intonation".

A man's conversational voice is assumed to be pitched at about "C" of 128 to 132 vibrations per second; and a woman's at two octaves higher, or 512 to 528, while a high soprano can reach high "C" of 2,048 to 2,112.

Some scientists have claimed that it is possible for some persons to hear and distinguish 11,000 tones ranging in pitch from 16 to 38,000 vibrations per second. Other than human life detects higher sound frequencies, but the limit of the average human ear is about 125 semi-tones of tempered scale intervals. Well-trained voices have a range of about 30 semi-tones (2½ octaves); the piano 88 semi-tones (7 1/3 octaves); the harp 78 semi-tones (6½ octaves); the cello and violin 48 to 53; and the symphony orchestra ranges 100 semi-tones (8 1/3 octaves). Most of the other instruments of the orchestra range about 30 semi-tones (2½ octaves), but the bassoon, clarinet, and bass tuba exceed three octaves.

Pitch

The frequency of tones identified by music notation as various pitches has varied from time to time in accordance with the way the scales have been formed and with the pitch designated for any particular note in such scale. Because of arithmetical simplicity the physicists have assumed a frequency of 256 for middle "C", and the "A" above would then be 426 2/3 by "just intonation". The French or International Pitch assigns 435 for "A", and the "C" below would be 258.65 on an "equally-tempered" scale; while U. S. musical instrument manufacturers and tuners have adopted 440 vibrations per second for "A", which would make middle "C" 261.6 for "equal-temperament", and 264 for "just intonation" in the keys of C, F, and B flat. But it is a matter of dispute whether it should hold such a value for any other key than that of G, wherein "A" changed its value from 440 to 445.5, depending upon the method of formation of "just intonation".

Mathematically, or physically, musical intervals are measured by the ratio of the frequencies of the notes, or some multiple of that ratio. But in music notation they receive the ordinal number (*e.g.*, third) corresponding to the cardinal number (three) from the lower to the upper of two notes (counting the lower as number one), when the scale is in the key of the lower of the two notes. The terms major, minor, augmented, and diminished are added to the ordinal number to more definitely describe the number of half tones included in the interval. When a major or perfect interval is widened by a semi-tone it is called an augmented interval; when narrowed by a semi-tone it becomes a minor interval; and when a minor interval is narrowed by a semi-tone it is called a diminished interval.

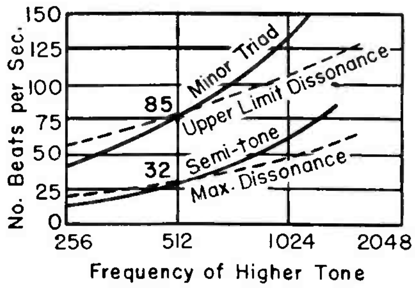


Fig. 1. Range of dissonance for middle frequencies.

If two tones are sounded simultaneously they form "harmonic" intervals; if sounded successively, "melodic" intervals.

The two-tone musical interval C to E is called a third, and that of E to G is also called a third; but these intervals are unequal because the former consists of four semi-tones in a tempered scale and the latter only three semi-tones; so the former is called a "major" interval and the latter a "minor" interval.

A three-tone chord is called a "triad", and it may be major or minor. A major chord is composed of a major third (4 semi-tones) above its lowest note with a minor third (3 semi-tones) above the second (or middle) note of the chord, thereby making the highest note a "perfect fifth" (7 semi-tones) above the lowest note. A minor chord is composed of a major third above the minor third which is based on the lowest note, and its total range of seven semi-tones makes the highest note also a "perfect fifth".

Consonant intervals are those which can go into the formation of a major or minor triad. Dissonant intervals are the major and minor second and seventh, and all augmented and diminished intervals. Whenever dissonant intervals enter into a chord, that chord is a dissonance; and it requires resolution into consonance to satisfy our normal human desire for stability and rest. Consonant intervals satisfy the ear because of the simple ratios of their frequencies; dissonances give a feeling of unrest and

TABLE II

CONSONANT INTERVALS			DISSONANT INTERVALS		
Order No.	Name of Interval	Frequency Ratio	Order No.	Name of Interval	Frequency Ratio
1	Unison or Prime	1:1	9	Major Second, or Whole Tone	9:8
2	Octave	2:1	10	Major Seventh	15:8
3	Perfect Fifth	3:2	11	Minor Seventh	9:5
4	Perfect Fourth	4:3	12	Triton or Augmented Fourth	45:32
5	Major Third	5:4		Harmonic Minor Seventh	7:4
6	Minor Third	6:5		Diminished Fifth	64:45
7	Major Sixth	5:3		Grave Minor Seventh	16:9
8	Minor Sixth	8:5		Minor Second, or Small Tone	10:9
				Semi-Tone	16:15

THE FREQUENCY RATIOS FOR INTERVALS AND TONES IN DIATONIC SCALES.

MAJOR SCALES (IONIAN MODE): CHORD RATIOS C:E:G:: 4:5:6.

Do or Ut — Re — Mi — Fa — Sol — La — Si — Do, Ut

C 9/8 D 10/9 E 16/15 F 9/8 G 10/9 A 9/8 B 15/8 C'

$\underbrace{1 \quad 9/8 \quad 5/4}_{5/4} \; \underbrace{4/3 \quad 3/2}_{6/5} \; \underbrace{5/3 \quad 15/8 \quad 2}_{4/3}$

ORIGINAL OR NATURAL MINOR SCALES (AEOLIAN MODE): A:C:E:: 10:12:15

La — Si — Do — Re — Mi — Fa — Sol — La

A 9/8 B 16/15 C 9/8 D 10/9 E 16/15 F 9/8 G 10/9 A'

$\underbrace{1 \quad 9/8 \quad 6/5}_{6/5} \; \underbrace{27/20 \quad 3/2}_{5/4} \; \underbrace{8/5 \quad 9/5 \quad 2}_{4/3}$

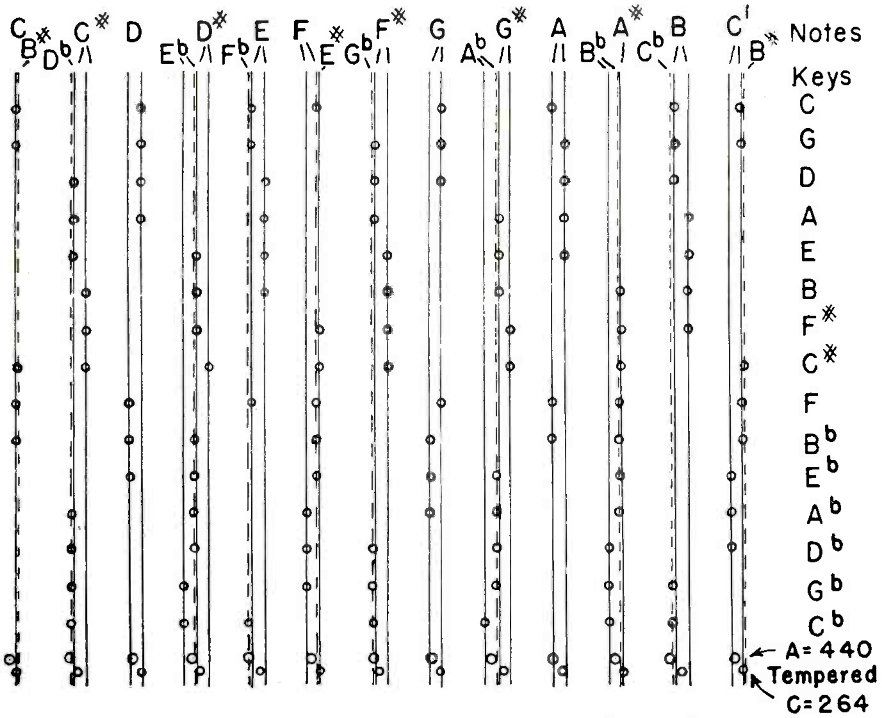


Fig. 2. Relative spacing of tones necessary for the fifteen major keys.

desire for resolution into consonance because of the complexity of such ratios of tone frequencies.

Discord

The sensation of discord is partly due to the "beats" of low frequency resulting from the summation algebraically of the two wave trains of slightly different frequencies (chiefly their difference), and so at some nodal points greatly augment their amplitude and at others tend to neutralize and make the sound very weak. The ensuing fluctuating pitch of the note (like "wow") adds to the unpleasantness. Such "beats" can be formed by the interference of the upper partials (harmonics) as well as by the fundamentals.

The amount of dissonance varies with the pitch of the tones for any given number of "beats", and is greatest for the following values: 20 to 23 beats for a frequency of 256; 32 for 512; 50 for 1024 cycles per second. The range of dissonance for the above pitches is, respectively: 1 to 60 beats; 1 to 85 beats; and 1 to 105 beats. See *Fig. 1.*

The first or key-note of a scale is called the "tonic"; and the following tabulation gives the ratio of the number of vibrations of the upper note of an interval to the tonic for both consonant and some dissonant intervals found in the natural harmonic series. In it the order of consonance decreases as its numbered position increases.

The ratio of a major to a minor tone is 81:80; and of major to minor thirds, sixths, and sevenths is 25:24. The ratios of major chords are 4:5:6:8; and of the minor chords 10:12:15:20.

To correspond to the frequency of the minor third of A-C in the minor scale, the interval of the major third C-E must be lowered by an intermediate tone which is called E flat in the major scale of C; and the ratio of the frequencies of E to E flat must be 25:24. Correspondingly, minor intervals must have added sharps to raise them to major intervals: and the sharps may be 1/24 higher.

Just Intonation

In this fashion, and with the arbitrary assumption that such fixed ratios applied to all intermediate notes, musicographers foisted tonal confusion upon the listening world years ago; and we have not recovered from it. They called such a scale "just intonation" and "natural intonation", both in error because its frequencies do not match the frequencies of the natural harmonic overtones of any tone.

But since there are three sizes of intervals in a diatonic scale, two of which have two as a factor and the other the factor three, no such constant factor could properly be applied for their reduction or increase to form the "chromatic scale", which has intervening sharps and flats.

Furthermore, since both whole tone intervals (1/8 and 1/9) were greater than twice 1/24 (equal to 1/12) they were led to the wrong conclusion that a sharp was usually less in frequency than the flat of the diatonic tone above it. For the exceptions the two so-called semi-tones (although in fact not half the value of any interval) with an interval of 1/15, so less than 1/12, caused E sharp to be higher than F flat, and B sharp higher than C flat. In such a manner the twenty-one tone chromatic *Scale 1* of *Table 1* was dubbed "just intonation". The frequency ratios of the equally-tempered scale are shown for comparison in the same table.

However, the ratios of frequencies of the tones in the seven-tone diatonic scale seem well established in all scales of just intonation, although their absolute frequencies may not be the same in all keys. The discrepancies between proposed chromatic scales of just intonation occur in the assignment of values for their intermediate tones, called "accidentals". The suitability of such a name seems justified by their dependence upon the whim of the scale designer.

In Grove's "Dictionary of Music and Musicians" appears under the title "Intervals" algebraic formulae for the construction of intervals from the frequency ratios of three stable intervals: the octave (2:1), the fifth, called quint (3:2), and the third, called tierce (5:4). That seems reasonable and logical until it appears that the frequency values thereby found for the key of C do not necessarily hold for the scales of other tonic keynotes. Therein, the method of arriving at the frequency ratio of any tone is motion forward and backward by the proper combinations of octaves, fifths, and thirds to arrive at such a note from the fundamental of C. The numerator and denominator of such ratio seem to correspond to the numbers of the natural harmonic partials. To express the interval in "cents", the logarithm to base ten of the ratio is multiplied by 3,986. A "cent" is one twelve-hundredth part of an octave, so that each semi-tone of an equally-tempered scale is represented by hundreds of "cents". *Scale 2* of *Table 1* shows Grove's intervals.

In Baker's "Dictionary of Musical Terms" apppear a few variants in the frequency ratios given in Grove's, for example: the augmented prime C sharp to C is given as 135:128 (1.05469); the augmented fourth F sharp to C as 25:18 (1.38889); the distinguished fifth G flat to C as 36:25 (1.44). The two latter intervals are based upon the mathematical assumption that the frequency of the higher tone is 25/24 that of the lower tone. It is inconsistent in not using such a ratio for the augmented prime although it does appear for its inverted interval C sharp to C', the diminished octave, as 25:48, which would correspond to an augmented prime of 25:24.

We find that the natural harmonic frequency ratios do not result in the same frequencies for the same name of note as the keys are changed. For example, in the key of C the note A has a frequency ratio to C equal to 1.6667, but in the key of G its ratio to C is 1.6875; similarly, E in the key of D is 1.26562 instead of 1.25 in the key of C; B in the key of A becomes 1.89844 instead of 1.875; D in the key of F becomes 1.1111 instead of 1.125 in the key of C; G in the key of B flat becomes 1.48148 instead of 1.50; F in the key of A flat becomes 1.31687 instead of 1.3333; and even C in the key of E flat becomes 1.97531 instead of its octave value of 2.00 in the key of C.

Therefore, the following progressively formed *Scale 3* of *Table 1* is proposed as the most reasonable chromatic scale; and the comparison is shown for its frequency values and those of *Scales 1* and *2*. In this tabulation 35 relative tone frequencies are shown as necessary in any octave of the chromatic natural harmonic *Scale 3* for the reproduction of all major and their relative minor keys: and the keys are indicated wherein each frequency is required. The derivation of its actual pitch frequency for the octave above middle C of 264 cycles per second is shown later in the discussion.

If each successive major key has its tonic (fundamental) a perfect fifth (usually called 3½ tones) above the tonic of the preceding key we find that an additional note is sharpened in each succeeding key, so additional sharps are added to the signatures; and the succession of keys is thereby C, G, D, A, E, B, F sharp, and C sharp. Similarly, if the tonic of each successive key is a perfect fifth (3½ tones) lower an additional note is flatted in each successive key and signature; and the series of keys becomes C, F, B flat, E flat, A flat, D flat, G flat, and C flat.

The construction of such keys is facilitated by the additions of the logarithms of the frequency ratios. The plotting of the resultant logarithms gives a graphic picture in *Fig. 2* of the relative spacing of the tones necessary for the fifteen major keys. The constant large deviations in frequency for any note correspond to what is known as the Didymic Comma (81/80), or one-eightieth of the lower frequency. The constant small deviations between the frequencies of notes which are called "enharmonic" (D sharp and E flat considered the same tone in a tempered scale) causes the sharp of the lower note to have a frequency 0.1128 percent higher than the flat of the upper note. This denies the oft-repeated dictum of musicologists.

Of the tones in *Scale 3* ten pairs have a slight deviation which, if ignored, would permit an octave of 25 tones, which is truer to a just intonation than the 21 tone octave usually so labeled. It should be remembered that such scales are organized consistent to the

[*Continued on page* 41]

NEW PRODUCTS

PARA-FLUX REPRODUCER

A new line of Para-Flux reproducers is now being manufactured by Radio-Music Corporation, East Port Chester, Conn. These reproducers are used by manufacturers of transcription consoles, by AM-FM broadcast stations, recording studios, wired music companies and theatres and factories that utilize recordings.

The three types of reproducers are for vertical only, lateral only, and universal uses. All three types are interchangeable with the Model A-16 arm and Model EL-1 equalizer illustrated above.

Vertical Head Model VL-1DA: Provides maximum in quality of reproduction from vertical ("hill and dale") recordings, while discriminating against lateral response. The diamond stylus has a 2 mil. radius.

Lateral Head Model LL-1DA: Designed to provide the most advanced quality in lateral reproduction, while discriminating against vertical response. Stylus has 2.5 mil. radius.

Universal Head Model UL-1DA: For superior performance where both lateral and vertical reproduction is required from the same unit. Selected diamond stylus has 2 mil. radius. Design is such as to permit useful output should lateral transcription be started while equalizer is accidentally left in a vertical switch position, and vice-versa.

Response of all models—linear from 40 to well beyond 11,000 cycles per second.

Plug-in Head: Vertical, lateral and universal heads use same arm and equalizer. Head can be removed and replaced in a few moments by virtue of plug connection.

Designed for Cuing: "Hair-line indicator on head and precise stylus construction make accurate cuing possible and permit "back-tracking" without damaging parts.

A new Bulletin PR4 fully describes and illustrates Para-Flux Reproducers . . . mailed upon request.

PICKERING EQUALIZER

Providing full compensation for most of the commonly encountered recording characteristics, the Model 163A Equalizer has recently been announced by Pickering & Co. It is designed to connect between the 500-ohm Pickering pickup and a flat amplifier, and is equipped with five degrees of equalization. Two positions provide for extended high-frequency response, giving flat reproduction to over 10,000 cps, with a choice of bass compensation of either 20 or 15 db between 500 and 50 cps. A third position provides complete compensation for NAB or Orthacoustic recordings, while the two remaining positions provide a 5,000 cps low-pass filter for the more effective reproduction of shellac records or worn transcriptions, with a choice of between 20 and 15 db of bass compensation.

Instantaneous recordings and most vinylite pressings in good condition may be reproduced effectively with the equalizer in position 1, with some improvement noted on over-bass records in position 2. Positions 4 and 5 give the same choice low-frequency compensation accompanied with a reduced high-frequency response for the elimination of surface noise. Position 3 gives the best results on standard transcriptions. This equalizer has an output impedance of 500/600 ohms, but transformers are available to match it to 30, 150, and 250 ohms, or direct to single or push-pull grids.

It is completely enclosed in a shielding case 3½ by 3¾ by 5 inches, and is mounted by two 8/32 screws on 1½-inch centers, with the knob shaft between them.

Curves and further information can be obtained from the national sales office, Pickering and Co., 29 W. 57th St., New York 19, N. Y.

Z-ANGLE METER

A new instrument in the field of electrical and electroacoustic measurements is announced by Technology Instrument Corporation. This instrument is known as the Type 310-A Z-Angle Meter and is designed for impedance and phase angle measurements over a frequency range of 30 to 20,000 c.p.s. Balance is accomplished by means of a single dial, which reads directly in impedance, and the phase angle of the unknown impedance is indicated on the direct-reading meter. Supplementary scales are provided on the meter to indicate the D of condensers and Q of inductors and means are provided to indicate a leading or lagging phase angle. The range of the instrument in impedance is from .5 to 100,000 ohms and is accurate to within ± 1% over the greater portion of this range. The phase angle readings are accurate to within ± 2 degreees. In terms of C, L and R,

respectively, the instrument has an effective range of 1,000 $\mu\mu f$ to 10,000 μf, 5 microhenries to 500 henries, and .5 to 100,000 ohms.

Technology Instrument Corporation reports that this instrument is available for prompt delivery. A copy of their new 8-page bulletin describing the instrument, its applications and principles of operation, may be obtained by directing an inquiry to the company at 1058 Main Street, Waltham, Mass.

NEW UNIVERSAL MICROPHONE

Universal's new cartridge type carbon microphone, the A174, is no larger—and considerably lighter in weight—than an ordinary pocket watch. This compact, lightweight size, together with highly sensitive response, recommends it for detectaphone, midget transmitter, low cost paging and call systems, and experimental and special uses.

The A174 is a newly styled version of Universal's former "W" model. It utilizes

single button construction and is fully insulated. Its impedance rating is 200 ohms and the output level is given as 12 db below 6 milliwatts for 100 bar signal. Despite its small size and light weight (¾ ounce!) the A174 is not a delicate performer but has plenty of stamina and is not troubled by heat or humidity.

Additional information is available from the manufacturer, Universal Microphone Co., Inglewood 2, California.

AUDIO DESIGN NOTES

THE CATHODE FOLLOWER

• The internal impedance of a cathode follower, "looking across" the cathode resistor, can be computed from the well-known relation

$$Z_k = \frac{R_k R_p}{R_p + R_k(\mu+1)} \qquad (1)$$

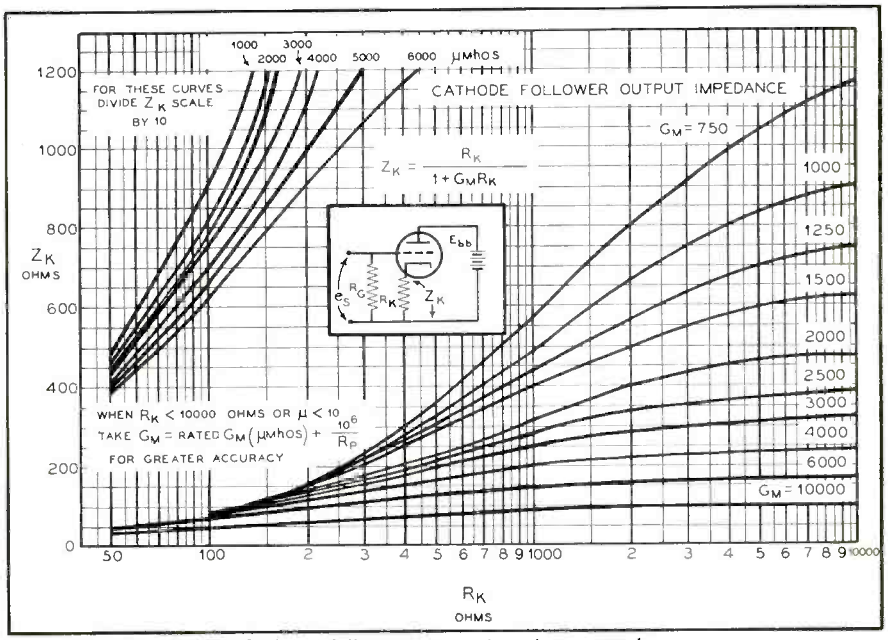
It would be more convenient, however, to have a graph from which the output impedance could be accurately determined for any value of R_k and a given set of tube parameters. The approximate relation,

$$Z_k = \frac{R_k}{1 + g_m R_k} \qquad (2)$$

is suitable for plotting such a graph. This has been done for values of g_m between 750 and 10,000 μmhos and values of R_k between 50 and 10,000 ohms.

The Z_k scale covers a range of 0-1200 ohms. For values of Z_k less than 120 ohms, the smaller family of curves applies if the Z_k scale is divided by 10. Values of g_m other than those plotted can be interpolated. In using the curves any two of the three quantities Z_k, R_k and g_m can be assigned and the third read off directly. For example, if the designer should want $Z_k = 500$ ohms, it is immediately evident from the curves that any value of R_k between 800 and 10,000 ohms, with corresponding values of g_m between 750 and 1850 μmhos may be used. Specifically, $g_m = 1500$ and $R_k = 2000$ would do the job. In the case where R_k is determined by the grid bias needed, say $R_k = 1000$ ohms and $g_m = 2600$ μmhos, Z_k is read off as 280 ohms.

Although the curves are plotted from the approximate *equation 2*, it is possible to get exact results if desired by adding a simple correction to g_m. It can be shown that Z_k as determined from *equation 2*, will be equal to Z_k as determined from *equation 1*, if a value of transconductance $g_m' = (g_m \text{ μmhos} + 10^6/R_p)$ is used in *equation 2*. It is absolutely necessary to apply this correction when μ and R_p are low, no matter how large g_m happens to be. For the error to be less than 5%, R_p must be greater than 10,000 ohms or μ greater than 10. In general the curves can be used with the rated g_m of most tubes and yield sufficiently accurate results. If the values of μ and/or R_p are not high enough, it is a simple matter to apply the correction to g_m and get accurate results with the curves.



Cathode follower output impedance graph.

Power Output

In some applications, such as driving a loudspeaker, it is required to supply considerable power under optimum conditions. The old rule of matching generator to load impedance holds true and can be readily applied. If R_k is the load resistance, maximum power transfer will occur when

$$R_k = \frac{R_p}{(\mu+1)} \qquad (3)$$

This is because $R_p/(\mu+1)$ represents the impedance "looking into" the cathode circuit of the tube. It is this impedance in shunt with R_k that represents the output impedance of the cathode follower.

The signal power in the cathode resistor can be shown to be

$$P_o = \frac{\mu^2 e_s^2 R_k}{2\,[R_p + R_k(\mu+1)]} \text{ watts} \qquad (4)$$

where e_s is the input peak voltage. If the value of R_k, as given by *equation 3*, is substituted in *4*, the result is

$$P_o = \frac{\mu^2 e_s^2}{8\,R_p\,(\mu+1)} \text{ watts max.} \qquad (5)$$

These two equations give the output power for any cathode load or for the optimum load. Their usefulness is considerably enhanced by an equation relating the signal voltage e_s to the actual grid to cathode voltage of the tube. Such

[*Continued on page* 45]

PROFESSIONAL DIRECTORY

TECHNICANA

CORRECTIVE NETWORKS

● The use of constant-resistance networks for equalizer circuits has become almost universal in the motion picture industry. While some use is made of them in original recording, they have still greater use in rerecording operations.

Two types of networks are in general use: those designed for a particular application, such as for dialogue equalization. film-loss compensation, and requiring a total of three attenuators when the device is to be usable without changing the over-all sound level.

The original design data for constant-resistance equalizers was contained in a previous article, by the same author, which appeared in the *SMPE Journal* of Nov. 1938. Together, these two articles provide considerable design data which should be maintained in the files of anyone who is ever faced with the necessity of designing equalizers.

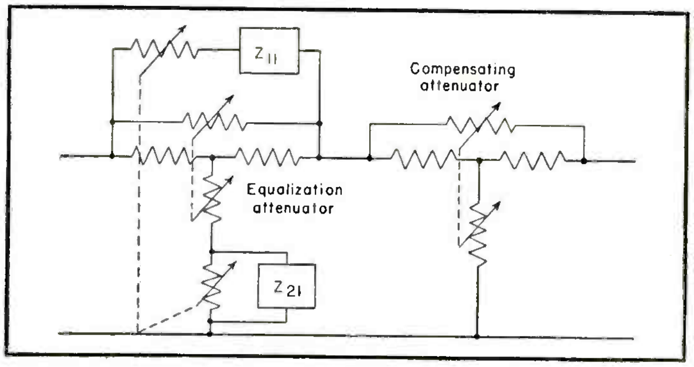


Figure 1

modulator compensation; and those which are for corrective use, as determined by aural monitoring, to provide improvement in quality or balance, or to produce required dramatic effects.

In the March 1947 *SMPE Journal*, F. E. Hopper describes some of the basic factors in the choice of equalizers, the effect of source and load impedances, and the use of compensating attenuators ganged with the equalization control so as to provide a change in the characteristic without a change in the over-all sound level.

Since a variation in the amount of attenuation used with a given pair of elements causes a variation in the shape of the resulting equalization curve, one additional control is necessary if it is necessary to maintain the same curve shape. For example, an equalizer designed to have a maximum loss of 10 db will have a certain characteristic at this loss, and at other steps of the controlling attenuator, the points of deviation from the flat portoin of the response curve differ from those existing when the equalizer is operated at the design point. This condition can be corrected by adding variable controls to the reactive elements, as shown in *Fig 1*, thus

PRE-EMPHASIS and DE-EMPHASIS

● Simplifying the calculation of a response curve corresponding to a pre-emphasis expressed in microseconds, an article by "Cathode Ray" in the May 1947 issue of *Wireless World* describes the terminology and outlines the methods used almost in words of one syllable.

When a pre-emphasis is expressed in microseconds, the response curve may immediately be imagined by those versed in this terminology. However, those who can only visualize a curve expressed as a boost of n db per octave may not be as quick in interpreting what is meant. Reduced to its simplest terms, a boost of 16 db is obtained at the frequency at which one cycle requires to the delay expressed in microseconds, or in other words,

$$f_h = 1/t$$

where f_h *is* the frequency at which the boost is 16 db, and t is the delay in microseconds. A second frequency of interest in $f_h/2\pi$, at which the boost is 3 db. Since the slope of the straight portion of the curve is 6 db per octave, it is possible to construct a fairly accurate curve with these two frequencies known.

For the converse calculations, to obtain the size of capacitor across a circuit of given impedance to produce a de-emphasis equivalent to a delay in microseconds, the following formula holds:

$$C = t\mu\text{sec}/R$$

where C is in microfarads and R is in ohms.

This will give the proper size of capacitor C to be used across a plate resistor R as shown in *Fig. 2*, to give a

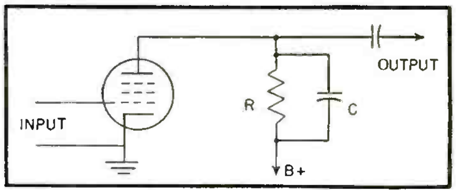


Figure 2

de-emphasis which is exactly complementary to the boost given by a pre-emphasis as expressed in μsec. In such circuits, however, it is necessary to include the effect of the following grid resistor.

REDUCING MICROPHONICS IN TRIODES

● An analysis of patent number 2.-389,935, described as a triode whose plate current varied by a displacement of the grid structure has been published by A. H. Waynick in the *Journal of Applied Physics*, February 1947. The author suggests that tubes be designed (when specifically designated as audio frequently amplifiers) on the basis of the term

$$Mp = \frac{d\ \text{(plate current)}}{\text{unit grid displacement}} = 0$$

where Mp is the motion factor of the grid structure.

It is pointed out that it is electronically feasible to obtain a plate current variation from a displacement of the grid in the triode. Since most microphonic difficulties arise from this mechanical motion of the grid it should be possible to consider this factor further in the construction of the tube. In laboratory tests conducted by the author, it was possible to vibrate the envelope of a sample 6A3 and although the grid bias and plate voltage were stable, the plate current indicated that the sample tube had seven dominant microphonic frequencies for the transverse and six microphonic frequencies in the longtiudinal between 100 and 10,000 cycles.

RE "CATHODE-COUPLED TRIODE AMPLIFIERS"

It is suggested that U. S. Patents 2,269,417 and 2,276,565 be added to the bibliography of C. J. LeBel's article, last month. Both are by Murray G. Crosby.

Audio Amplifier

[*from page* 9]

Transient waveforms should be transmitted without distortion since audible sounds have very important transient characteristics. Percussion instruments and staccato scores on the brasses demand good transient response. A square wave introduced into an amplifier and viewed on a scope is a good test for transient responses. *Fig. 3* shows a 5000-cycle square wave through this amplifier.

Good transient response is accomplished by extending the frequency range and having negligible phase shift between input and output of an amplifier.

Transformer resonance can cause slight oscillations well above the audible range. If the phase shift at these oscillation frequencies is sufficient to cause the feedback to become positive, regeneration will take place, resulting in sustained oscillations and overloading of the amplifier at super-audible frequencies. This can be avoided by reducing the gain of the amplifier above the useful frequency range, or by preventing the occurrence of these oscillations by proper design of the output transformer.

The transformers are completely free from saturation or leakage reactance effects from 25 to 20,000 cycles at any power up to maximum output. The low frequency response is flat within 2 db to 3 cycles. The extremely low frequency response in the amplifier is attenuated in the first stage to eliminate the effects of transients in tuning a radio receiver, or phonograph turntable eccentricity, or rumble. Some amplifiers for industrial application have made use of this good low frequency response. The low-frequency attentuation is accomplished by using a condenser in series with the input to the grid of the first tube. The complete schematic is shown in *Fig. 4*.

The output transformer is largely responsible for the fine performance of the amplifier. It looks like a simple endbell type of transformer, but with larger-than-the-usual output transformers. The amount of iron in the transformer and the unique, very complicated wind structure gives the transformer no frequency discrimination and negligible phase shift over the entire audio range.

Negative Feedback

Everything possible was done to produce a perfect audio amplifier without using feedback. After this was accomplished, 11 db of feedback was added. This amplifier without feedback has as good a frequency response as usual high-fidelity amplifiers with feedback. In other words, it does not depend entirely upon feedback for its 0.2 db variation from 20 to 20,000 cycles.

The input impedance to the amplifier is normally 0.5 megohms. An input transformer is hermetically sealed. It is well shielded from magnetic pickup by having three nickel alloy shields. The leads pick up more hum than the transformer.

Power is available for operating additional preamplifiers or a tuner. The voltages available are 6.6 volts at 5.5 amps, and 300 volts at 90 ma, d.c.

The distortion at full output of 30 watts is less than 2½ per cent. The intermodulation distortion, or double-frequency results, are extremely low. Taken at 24 watts at 50 cycles and 1.5 watts at 1000 cycles, the intermodulation distortion is 1.7 per cent. Taken at 4.7 watts at 50 cycles and 0.3 watts at 1000 cycles, the intermodulation distortion is 0.2 per cent, referred to the 1000-cycle signal.

Listening tests carried out in conjunction with a wide-range loudspeaker system have fully supported the measured performance. We could not detect distortion in reproducing organ music including 25-cycle pedal notes. It provides an ideal amplifier for sound-recording purposes, FM monitoring or anywhere else where distortionless amplification is necessary.

Musical Acoustics

[*from page* 37]

progressive series of tetrachords employed by the ancient Greeks, with the tonic of each successive key having a frequency the same as the perfect fifth higher or lower than the immediate preceding keys. The minor keys have a tonic a minor third less than its relative major key, whose signature it bears.

(*To be continued*)

Perfect Reproduction

[*from page* 27]

other orchestral instruments, of having strong upper partials in the lower and middle ranges, while the higher notes tend toward pure tones.

The advantage (or otherwise) of reducing the upper end of the frequency scale can be checked by the reader in a very simple manner when listening to an actual orchestral performance, preferably when listening to the performance of a brass band in the open air. The acoustic output of instruments like the trombone and trumpet is largely

concentrated along the axis of the horn, the concentration becoming more marked at high frequencies. Thus the frequency characteristic is "flat" on the axis, while the high frequency output falls relative to the 500 cycle components as the listener moves off the axis, *i.e.,* "top cut," can be put in without electrical equipment being involved. If this experiment is tried indoors the listener should be fairly close to the instrument in order to minimize the effect of reflections.

The theme will not be pressed further, but instead an attempt to draw some conclusions will be made.

It appears that in a high quality reproducer system with total distortion 10-15 db below the level at which separate tests indicate distortion as "just noticeable," public preference for a restricted frequency range exists. This optimum range is approximately 3 db down at 70 cycles and 6,500 cycles, but is somewhat dependent on program material.

A similar preference appears to exist when electrical reproduction is not involved, except that an even more restricted range appears to be preferred, though absolute data on this point is not as complete as is desirable.

It does appear that the presence of distortion is not the only reason for the general desire to "cut top."

In providing a tone control the engineer is making it possible for the great mass of the general public to express an opinion on a subject that has previously been the prerogative of musical circles. It is suggested that this opinion serves to confirm the general trend of the tone quality of orchestral instruments, though further investigation is required.

Insofar as the public tends to go further than musical instrument designers have gone in removing the upper partials it would appear that some organized investigation of public preference is necessary as a guide to the designers of musical instruments.

Assuming the correctness of the present thesis it may be well asked, "What is the communication engineer going to do about it?" It is suggested that the present practice of transmitting with a flat or slightly rising characteristic and providing each listener with a tone control is the correct procedure. This gives a measure of pre-emphasis and de-emphasis, which, while not going as far as American practice goes (wrongly, in the writer's opinion), does provide a useful improvement in signal/noise ratio.

As to the technically minded listener, it is suggested that he forgets his preconceived ideas on where the tone controls should be set and adjusts them to give the most pleasing reproduction. To save the conscience it would be a good idea to keep a log of the settings found most pleasing with different types of programs, and after about a year try and analyze the results with a view to ascertaining the preferred frequency characteristic.

In presenting this information on such a widely debated subject, the writer has attempted to avoid inserting any personal bias either for or against the subject matter. The results present a disconcerting discontinuity in our previous line of thought and it would appear imperative to obtain some further evidence of the mass reaction to frequency range restriction. The individual opinion of an engineer would appear to be less important than that of an ordinary member of the public lacking all knowledge of the prior art.

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Olson Report

[*from page* 27]

quality and pronounced it correct and the balance true.

Listeners were picked at random and all common occupations and both sexes were represented. Tests were made by playing two complete selections, switching filter setting about every fifteen seconds. Listeners then checked *A* and *B* space on card, listed occupation and comments.

Over 1000 listeners were used in tests with popular music, with the following result:

Prefer full range 69%
Prefer restricted range 31%

This could be broken down into various age groups:

	Per Cent Preferring	
Age	*Full Range*	*Restricted Range*
14 — 20	59	41
20 — 30	67	33
30 — 40	75	25
40 — 65	69	31

Tests using semi-classical music were made, with about 200 listeners, with this result:

Prefer full range 66%
Prefer restricted range 34%

Other tests were made using speech,

[*Continued on page* 44]

with no less perference being shown for full range.

It is entirely evident that users prefer full frequency range when dealing with a system with no distortion. This test is only the beginning of a broad analysis of user preference. Particular attention will be given to the effect of distortion. It is certain that the conditioning of the public is not predominant, and that the tone of present musical instruments is not fatally defective.

In opening the discussion someone repeated a quotation from Professor Bolt's remarks at an earlier session, that the public had had many centuries to forget musical instruments of displeasing tone.

There was a question as to the divergence between these results and those of other widely discussed recent studies. J. P. Maxfield, of Bell Telephone Laboratories, arose and welcomed studies such as these; he had found that with electronic systems which were free from high frequency transients and from cross-modulation, listeners preferred wide-band reproduction. This was true in both single channel and stereophonic systems. There was some preference for diffuse sources.

John P. Taylor said that with an ordinary system the public wanted good signal to noise ratio above many other considerations.

There was some discussion of the possibility of transient distortion in the low-pass filter used in these tests. Dr. Olson said that the filter terminated in its characteristic impedance, and that it had a lower Q than had the musical instruments, hence transient effects should be negligible. Mr. Maxfield said that a sharp cutoff was not harmful if the attenuation increased continuously to inaudibility (as was the case here). A filter in which the attenuation increased rapidly to say 50 db, then decreased to say 40 db and leveled off, was very bad: squeals and ringing always resulted.

Mr. Dunbar wondered whether reverberation might not be excessive in the corner with the filter closed. Dr. Olson pointed out that the filter was acoustically transparent in the range it was transmitting and there could be no "boxed in" effect.

This concluded discussion on the paper.

EASTERN AMPLIFIER APPOINTS

Burlingame Associates Limited, 11 Park Place, New York 7, New York, have been appointed to represent Eastern Amplifier Corporation, 74 East 140th Street, New York 54, New York, in the New England States, Metropolitan New York, New Jersey Eastern Pennsylvania, Maryland, Delaware and District of Columbia.

Record Revue

[from page 33]

will shortly be overcome in one way or another because the finer performance of these new units, once heard, is not likely to be forgotten by those who want the best in reproduction.

Here are some recent albums with outstanding quality of reproduction:

Brahms—Sonata in F Minor, Op. 120, No. 1;
William Primrose, viola William Kapell, pianist .... **Victor M1106**
(3 records)

Surfaces are not too good, but there are more highs in this than is usual. "Liveness" balance brings viola too close, piano seems relatively distant.

Tschaikowsky—Symphony No. 2 ("Little Russian")
Minneapolis Symphony Orchestra; Mitropoulos .... **Columbia M573**
(5 records)

Another of the wide-range orchestra recordings Columbia has recently been issuing. This one was evidently recorded some time before most current releases, but it matches them in quality.

Khachaturian—Masquerade Suite
Kabalevsky—Fête Populaire.
Santa Monica Symphony Orchestra; Rachmilovitch .... **Asch 800**
(3 records)

A brilliant high-fidelity, studio-made recording. Good example of difficulty with too much deadness. Highs evidently pre-emphasized. Very heavily recorded. Liveness added artificially would help this recording musically.

Humoresque (From the film). Isaac Stern, violin. Orch. conducted by Waxman .... **Columbia M657**
(4 records)

A beautifully done, concerto-style recording, with perfect balances and liveness treatment for the solo violin. Music is re-composed classics, highly doctored.

Kauder—Sonata for English Horn and Piano.
Louis Speyer, English horn; Erwin Brodky, piano .... **Night Music 105**
(2—10" plastic records)

A new company issues a remarkably high-quality recording of the rich sound of the English horn. Very quiet surfaces, excellent fidelity. Music is nothing exceptional.

ATLAS APPOINTS BRESSLER

Atlas Sound Corporation of Brooklyn announces the appointment of Jules J. Bressler as field sales engineer for the New York metropolitan area and northern New Jersey, for the industrial and jobber trades.

Mr. Bressler is well known in the motion picture and radio fields, and has been actively engaged in the sales engineering of sound and associated equipment for the past twenty years. He was Chief Engineer of Beltone Sound Systems Co. for several years. His technical training in these fields are available to his many contacts on whom he will call.

Audio Design Notes

[*from page 39*]

an equation is,

$$e_s = \frac{[R_p + R_k\,(\mu + 1)]\; e_g}{(R_p + R_k)}$$

For the condition of maximum power output using optimum R_k, *equation 6* reduces to,

$$e_s = \frac{2\;(\mu + 1)\;e_g}{(\mu + 2)}$$

These equations take account of the degeneration and allow the designer to determine the input voltage, having been given the grid voltage or vice versa. It is left to the reader to write his own transconductance forms of *equations 3, 4, 5, 6* and *7*, keeping in mind their approximate nature when μ and R_p are low.

Practical Dividing Networks

[*from page 17*]

phase. In operation, this test unit is set for transmission to the low-frequency speaker, and the acoustic output at the crossover frequency is measured using a microphone and amplifier with an output meter. The switch is then thrown to the high-frequency position and the pot is adjusted to obtain the same acoustic output from the high-frequency horn. Then, with the switch in the center position, both speakers are energized at the crossover frequency and the DPDT thrown from one position to the other to obtain the maximum output as measured by the microphone and amplifier. Fine adjustment of phasing is then made by moving the high-frequency unit and horn back and forth relative to the low-frequency unit to obtain the maximum output from the combination. The relative positions of the two speakers are then noted and final assembly of the speakers is arranged to maintain these positions. If maximum output should happen to be obtained with the high-frequency horn extended beyond the front of the low-frequency baffle, this condition may be corrected by reversing the DPDT switch and sliding the horn back one-half wavelength at the crossover frequency. *Table 1* gives values suitable for constructing the 1 db/step "scaling hook" pot for an impedance of 10 ohms. For any other impedances, Z, the values given should be multiplied by Z/10.

Such an elaborate device is not necessary, however, for making final adjustments to two-way speaker systems. The acoustic outputs of the two speakers can be balanced close enough by ear, using the tap on the resistor R_1 of *Fig. 1* for adjusting the relative levels.

TABLE I

10-ohm T-pad Resistance Values

Loss db	Series Arm (2 required)	Total Shunt Arm
1	0.58	86.7
2	1.14	43.0
3	1.71	28.4
4	2.26	20.9
5	2.80	16.4
6	3.32	13.4
7	3.82	11.2
8	4.30	9.45
9	4.76	8.13
10	5.20	7.02

Phasing

Phasing adjustments between the two units may be arrived at by listening, preliminary settings being obtained by a simple reversal of the leads to the high-frequency unit, and final positioning being determined by continued listening. It will be noted that when the optimum point is located, the maximum realism will be heard from a two-speaker combination. With crossover in the vicinity of 500 to 1,000 cps, a reversal of the leads will give the effect on speech of jumping back and forth between the two speakers, completely eliminating the illusion of a single source of energy.

Final adjustments on any two-way speaker system should be made with whatever screening material is to be used in place, for two reasons. Most important, there is certain to be some attenuation of the high frequencies, and this should be compensated by adjustment of the resistor R_1. In addition, however, the illusion of a single sound source is often destroyed when both units are separately visible, and the covering with suitable grill cloth will eliminate this effect and blend the two sources into one apparent source, provided the phasing is correct. It is suggested that no adjustments to a system of this type be considered final until the listener has "lived with" the speaker for several days, and it should be expected that gradual improvements may be obtained in the over-all performance for the first two or three months after its installation as the user becomes familiar with its characteristics.

With this simplification of dividing network design and coil construction, it is believed that the average high-fidelity enthusiast should be able to obtain reasonably good results with a minimum of equipment being necessary to make adjustments. A loudspeaker system is designed primarily as a medium for transmitting sound to the ear, rather than to a microphone and a group of measuring instruments, and while no deprecation to the value of measurements is intended, it must be realized that the ear is actually the final judge of the performance of any loudspeaker system. If the constructor is able to obtain satisfactory results from the information contained in this article, its function will have been fulfilled completely.

Cathode Phase Inverter

[*from page* 19]

as a grounded grid amplifier and the exciting voltage is due to the IR drop across R_K. The voltage gain from R_K to R_{L2} is

$$V.G._2 = \frac{i_2 R_{L2}}{R_K(i_1-i_2)} = \frac{\mu_1 e_s R_K(\mu_2+1)R_{L2}}{\mu_1 e_s R_K(R_{P2}+R_{L2})} \quad (30)$$

This reduces to

$$V.G._2 = \frac{(\mu_2+1)R_{L2}}{(R_{P2}+R_{L2})} \quad (31)$$

It will be noticed that R_K does not appear in equation 31 and therefore it is the same as the expression for the voltage gain of a single grounded grid amplifier with no cathode resistor. V_2, then, does not have any feedback in the normal operation of the basic circuit. However, since the grid voltage for V_2 is a function of the plate current of V_1, its output is indirectly influenced by the feedback on V_1. The amount of this feedback in most cases will be small, so that very little effect on distortion can be expected. On the other hand, the impedance looking back into the plate circuit of the inverter stage will be raised somewhat, particularly on the V_1 side. This will tend to compensate for the lower value of R_{L1}, which can be used to obtain balance in some forms of the design.

Application to the Power Output Stage

Other writers[1] have suggested that cathode phase inversion can be incorporated in the power output stage, thus effecting an economy of tubes and components. The possibilities of this arrangement are of sufficient interest to warrant a detailed analysis. Suppose, for a start, that a pair of tubes such as 2A3's or 6L6's were to be connected in the manner of *Fig. 3*, with the load resistors R_{L1} and R_{L2} replaced by the high windings of a conventional output transformer. Let the resistance R_K be set to give the proper self-bias and the signal voltage e_s be large enough to drive V_1 to a reasonable output. Under these conditions the voltages e_1 and e_2, across the transformer windings, would be equal because of the unity ratio existing between them.

[1] Cathode Phase Inversion, by O. H. Schmitt —*Journal of Scientific Instruments*—Vol. 15, pg. 100—1938.

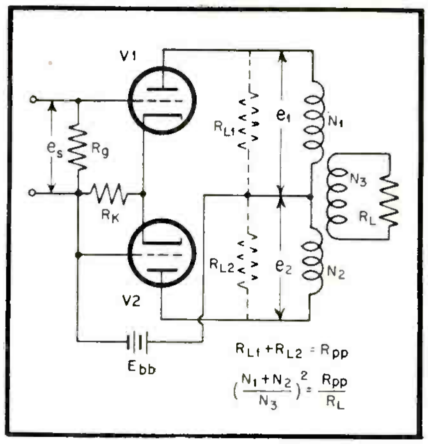

Fig. 3. Basic output circuit with cathode phase inversion.

However, the previously derived equations and a little logic will tell us that the grid voltage and plate current of V_1 are considerably larger than those of V_2. This means that V_1 is supplying a major portion of the output power and the circuit is badly unbalanced. The criterion in this case is obviously equal power output from both tubes and not equal voltages.

This condition cannot be brought about by increasing R_K indefinitely, because by the time some reasonable power balance is reached, R_K consumes too much of the plate supply voltage to be practical. In fact, for 2A3 tubes the R_K drop would exceed 1000 volts and for 6L6's would be over 400 volts, to achieve a 5-10% unbalance. The only practical way out of this difficulty is to design an output transformer that has an unbalanced secondary tap. This can be done so that each tube produces the same flux in the core and hence contributes the same power to the load winding.

In order to do this two conditions must be satisfied:

a) The total secondary should present the desired plate to plate impedance R_P
b) The N1 turns in both sections of the secondary, due to signal current, should be equal.

Condition (a) can be satisfied by having the total secondary turns give the proper impedance ratio referred to the primary. Condition (b) can be met by a procedure that follows.

From the basic circuit of *Fig. 3*, it can be stated that

$$R_{L1} + R_{L2} = R_{PP}$$

or

$$R_{L1} = R_{PP} - R_{L2} \quad (32)$$

By substituting this value of R_{L1} in equation 10, with $\beta = 1$, and solving for R_{L2},

$$R_{L2} = \frac{R_{PP}(1+g_{m2}R_K)}{1+2g_{m2}R_K} \quad (33)$$

The required values of R_{L1} and R_{L2} for an assigned total and balanced output are now established. Under these conditions,

$$\frac{i_1}{i_2} = \frac{R_{L2}}{R_{L1}} \quad (34)$$

Condition (b) may be written as

$$i_1 N_1 = i_2 N_2$$

from which

$$N_2 = \frac{i_1}{i_2} N_1 \quad (35)$$

As shown in *Fig. 3*, N_1 and N_2 are the portions of the high winding facing V_1 and V_2 and N_3 is the load winding. With both tubes in the circuit, under operating conditions, the impedance facing each tube will be proportional to the number of turns on its part of the high winding. This is because of the coupling between N_1 and N_2 and for a good output transformer represents a very close approximation. With this in mind it is justifiable to write

$$\frac{Z_{L2}}{Z_{L1}} = \frac{N_2}{N_1} = \frac{R_{L2}}{R_{L1}} \qquad (36)$$

From equations 34 and 35

$$N_2 = \frac{R_{L2}}{R_{L1}} N_1 \qquad (37)$$

It is also true that

$$\left(\frac{N_1+N_2}{N_3}\right)^2 = \frac{R_{PP}}{R_L} \qquad (38)$$

By substituting equation 37 in 38 and solving for N_1

$$N_1 = \frac{N_3 R_{L1}}{\sqrt{R_{PP} R_L}} \qquad (39)$$

The necessary relations have now been developed from which the turns ratios of the plate windings, relative to the load winding can be determined.

By designing the inverter output stage in this way it is possible to realize gains and efficiencies that approach those of a conventional push-pull circuit and partially obtain the cancellation of even order distortion products. This can be done with an economy of circuit elements and without the use of excessive plate supply voltage. A working example will be illustrated in Part II of this article.

The general theory of the cathode phase inverter has now been covered in some detail. Useful design equations have been developed as well as others that demonstrate particular qualities of the circuit. In the following section a few working examples will serve to illustrate their use in practical design procedure.

Ultrasonics

[from page 32]

or by evasions, that is, applying the forces in some new manner.

The use in processing is most promising. We are applying enormous forces to particles, breaking them up, stressing their boundary layers and affecting them in numberless other ways. Since most commercial things are made up of particles this means we have a chance to enter almost every field of manufacturing. We are dealing with the fundamental building blocks of materials, and are able to affect even large molecules. The published papers show that pure physicists with a few watts have uncovered several hundred highly desirable effects, and with higher powers and our eye on commercializing those effects, we should have a business in several years that will begin to be as important as electronics.

Simple RC Filters

[from page 29]

there is a recognized need for improvement, artificially, acoustically, or both. Few records are found (especially American) which have excessive studio reverberation, but most have too little. Many fine symphony albums have been practically ruined by recording in a studio which looked perfect but give the final playback the effect of having been recorded in a closet. Many European recordings are much more pleasing than American because of more natural reverberation, even though they may be inferior in some other ways. Every Sunday you can hear live symphony broadcasts which originate in a relatively dead studio.

High level groove modulation is a source of distortion on many otherwise satisfactory recordings. Here again some European recordings are superior. This is the natural result of extending the dynamic range and improving signal to noise ratio, but whether it is worth the distortion and rapid record wear which usually result is debatable. Some records in high-priced releases show practically no land between adjacent grooves, and when this occurs in grooves near the center of the record, the average pickup cannot be expected to track properly. Whether any pickup can properly reproduce such modulation is questionable because of the distortion resulting from the needle attempting to follow the groove, even with an ideal pickup.

When a record meets all the above requirements, as occasionally happens, the result is one which is technically unusual and may become a good prospect for demonstrations, especially if the artistry is good. These records are universally used by dealers to demonstrate their phonograph combinations, and are cherished by record collectors.

Some of the fundamental reasons for distortion may be shown graphically in a simple manner. The mathematical analyses are necessary for design improvements, but a simple way to illustrate the troubles encountered by the needle in following the groove is to use the imagination a bit. In cutting the master, it can be imagined that the stylus of a high quality cutter, moving in soft wax, can cut a groove which has good playback possibilities. Even

ADVERTISING INDEX

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neglecting the pitfalls of manufacturing the final pressing from this beginning, the needle of the pickup is expected to follow the groove modulation, even though it in no way resembles the cutting stylus. Apparently the cutting stylus must be plow-shaped to cut the groove, and the playback needle must be round to be practical. Whether these accepted shapes can be altered to be identical is difficult to say with finality. At any rate, the plow-shaped stylus makes a certain cut and the round needle in riding the groove gives a different motion. The difference between the two motions depends upon the amplitude of the modulation and the frequency. As the amplitude and frequency increase, the lateral swing of the needle varies more from the original cutter stylus swing, and the vertical component increases. The result is distortion at high frequencies and modulation, and is most apparent at the inner grooves where the groove speed drops, often making the tracking angle error of the tone arm seem more important than it is.

Graphical Representation

This may be shown in its simplest graphical form by using two sharp pencils. The two pencils are held together so both points touch the paper at the same time and form two parallel lines. Now if the pencils are moved so the lines drawn are separated by the same distance as the points, the lines drawn will simulate the groove cut by the cutter stylus. If the pencils are moved across the paper and deflected transversely similar to the stylus motion in the groove being cut, the lines drawn will roughly portray a modulated groove as viewed from above. With a little practice the lines can be varied so as to show different frequencies and amplitudes. By drawing rough sine waves the lines will be seen to come closer together at high amplitudes, which would "pinch" the round playback needle riding in such a groove. If the process seems worthwhile at this point, it may be carried a step further by adding a diameter equal to the distance between the pencil points. This circle represents the diameter of the playback needle. Now by superimposing the circle on the groove and moving it along the groove, it will be seen that the needle radius cannot follow too high frequencies, and that the needle is pinched by high amplitudes and high frequencies. This means that a smaller diameter is required at such times, and that the needle will be deflected vertically to find the smaller diameter, giving the vertical modulation.

To see what effect this has on the horizontal component, draw a series of concentric circles on the tracing paper, and at various points along the modulated "groove" fit the appropriate circle and mark the center through onto the paper below. The series of points made by the various centers show the curve of the horizontal movement of the needle. To compare this with the true modulation, trace one of the original outer lines, compare it to the other outer line to be sure they are identical, and then compare to the line of centers. The differences will depend upon the amplitude and frequency, and this distortion would be present with an ideal pickup.

This method is crude but gives a simple demonstration of lateral recording. It also helps visualization of the limiting factors of frequency response, and distortion at high amplitudes even though the cut does not interfere with the adjacent groove. No matter how these factors are demonstrated, they are present. Pickups which respond to vertical stylus motion (and many do) will give excessive surface noise and distortion. High frequency pre-emphasis will increase distortion.

Even with the above factors present, properly made recordings have a high entertainment value, but the factors make the future of disc recording questionable as far as competition with wire or some other form of recording is concerned. Possibly, when FM is popular, wide-range speakers and amplifiers widely used, and stereophonics practical, the present form of disc recording (vertical and lateral) will become obsolete. Maybe it will be considered a milestone like acoustic recording, but at present it is certainly good business if properly controlled.

Adapting Recorders

[*from page* 14]

development of editing machines similar to the moviola used by the motion picture industry.

Conclusion

It is expected that improvements in handling technique will parallel improvements in recorder performance. Paper-tape magnetic recording is in the broadcasting business to stay and has a growing field of application in this and other fields. It is felt that present equipment, properly used, is capable of handling many broadcasting assignments and that its current use will enable the broadcaster to obtain valuable experience in the use of the medium.

The adaptations described in this paper were made under the general direction of Mr. Howard A. Chinn, CBS Chief Audio Engineer.