AUDIO ACCESSORIES FOR SIDEBAND

RESTRICTING FREQUENCY RANGE IN TRANSMITTER AUDIO SYSTEMS

From July-August, 1949

TECHNICAL TIDBITS

RESTRICTING SPEECH RANGE IN SPEECH AMPLIFIERS

Note: ***The following article was prepared for publication before the April 27 FCC proposals regarding restricted bandwidth were made public. The attenuation of unwanted audio frequencies as discussed in this article is in the order of 12 db per octave. Because the FCC has given no details of the attenuation they consider necessary, there is no way of knowing whether 12 db per octave will be considered adequate.***

This is a case of where you can get something for nothing, or at least, close to nothing. Before giving the punch line, though, let's examine the situation from the beginning.

Phone stations on the ham bands seem to fall into three categories regarding their speech quality. The first are the stations that will have no audio equipment in the shack unless it is capable of a flat response from 20 cycles to 15,000 cycles. Their quality is superb, and your ears would tell you so if it were possible to have a receiver and a reproducing system capable of handling this audio range at a time when propagation conditions allowed undistorted reception. These amateurs are taking up needless space in the limited ham spectrum by their activities, but as long as their carrier is inside the band edge by twenty to twenty-five kilocycles (in order to keep those wide sidebands inside the band) then, the FCC will not bother them, at least not yet.

On the other extreme is the second group, small though it be. These amateurs wish to have a transmitter that is as effective, communication-wise, as possible. Those who are on AM phone tailor their speech amplifier equipment until it transmits the narrowest possible audio range, leaving only enough audio range for complete understandability. A more rabid group goes even further, by partially eliminating the carrier and then transmitting only one sideband. These amateurs deserve a lot of applause, but we needn't bother to applaud them, because they did this not for applause but because they want their money's worth out of their equipment.

Which brings us to the third group, which must certainly include the majority of the world's phone men. This group is made up almost entirely of Mr. Average Phone Man and others of his ilk. Mr. Average Phone Man has a speech amplifier and a modulator which he copied faithfully from some handbook or some radio magazine. When he finished the audio end, he connected it to his c-w rig, got on the air, and asked the first ham he contacted the age-old question "How's my modulation?" Aside from the fact that Mr. Average Phone Man should have checked his modulation with a scope, while transmitting into a dummy load, instead of depending on the advice of another Mr. A. P. M., this situation is quite normal and is to be expected.

All right, you say, this is old stuff, so where's the pitch? Here it is. Why continue to waste power by transmitting certain audio frequencies if these audio frequencies are unable to help the other fellow hear you, especially when you can almost get rid of these unwanted high and low frequencies at practically no cost? To be specific about cost, the change can be made by the use of four 600 volt paper or mica condensers.

Before explaining how and where to put which condensers, let's make certain that another point is clear. This article has nothing to do with speech compressors, speech clippers, or sharp cutoff low-pass filters. The latter will do an excellent job of tailoring the speech range, but these filters may be rather elaborate. Speech compressors and speech clippers, on the other hand, do not affect in any way the band-pass characteristics of an amplifier unit. They may, however, affect the fidelity from a distortion standpoint. This is especially true of speech clippers.

One other point might also be explained here. The changes to be described are suitable for practically any type of speech amplifier. However, a restricted bandwidth is not assured if these changes are made in an amplifier which is used for NBFM. If the swing is not carefully adjusted the bandwidth may still be excessive. In other words, it is worthwhile to make these changes in an NBFM speech amplifier, but the effect will be nullified if the signal is permitted to swing too far frequency-wise, due to improper adjustment.

Here, then, is what you may do to restrict the audio range of your speech amplifier in an economical way. First, attenuate the low audio frequencies by changing the value of two of the interstage coupling condensers and second, attenuate the high audio frequencies by adding a condenser from plate to ground on two of the audio stages.

The calculations to determine the proper size of condenser for each point are not difficult. It is first necessary to decide on the audio range you wish to cover. Let us assume that you want an audio characteristic which is down somewhat at 300 cycles on the low end and 3500 cycles on the high end. To be more exact, this is one which will be down 6 db at

300 and 3500 cycles, when changes are made to two of the stages. These two frequencies—300 and 3500 cycles—will be used in the calculations.

The next step is to examine the circuit diagram of your speech amplifier. Most amplifiers consist of a pentode preamplifier, driving a triode or pentode amplifier, driving a phase inverter or transformer coupled amplifier which in turn drives the output stage. We are interested only in the first two tubes. We want to put a condenser from the plate of the first tube to ground, and one from the plate of the second tube to ground. Also, we wish to change the values of the condensers which are between the plate of the first tube and the grid of the second tube, and between the plate of the second tube and the grid of the third tube.

If the third tube is a phase inverter, it is best not to attempt to change the coupling condenser between the second and third tubes. The reason is beyond the scope of this article but it might be necessary to change the grid circuit of the phase inverter in order to get the proper effect from the changed coupling condenser. In this case, the coupling condenser can be changed between the microphone and the input tube. This is completely satisfactory if a dynamic microphone is used. If a crystal microphone is used, a different approach is necessary. Again this is not within the scope of this article, so that you will have to be satisfied with changes on only one tube, instead of two.

The final step before starting the calculations is to check the value of the grid resistor to which the new coupling condenser will connect. This will be the grid resistor for the second and third tubes unless, as stated above, it is necessary to put one coupling condenser between microphone and grid, in which case examine the grid resistors for the first and second tubes. These resistors should be no greater than 250,000 ohms. If they are of a greater value, decrease them so they are 250,000 ohms or less. Incidentally, the grid resistor for the second tube is usually the gain control.

The proper value of coupling condenser will now be one whose capacitive reactance, at 300 cycles, is equal to the grid resistance in the grid circuit of the stage to which it connects. These words mean, simply, that the condenser value in micro-farads is equal to: $\frac{1{,}000{,}000}{(1884)\ (R_G)}$ where R_G is the value of the grid resistor in ohms. This assumes that the low frequency point selected was 300 cycles. The figure of 1884 is 300 times 2 times π. As an example, if the grid resistor is 250,000 ohms, the condenser should be 0.0021, so use a 0.002 mf condenser. Make this calculation for both stages, and replace your present coupling condenser with the calculated value of condenser if it is not already that value. The low frequency audio tones are now taken care of.

Before starting the calculation of the plate to ground condensers, find out the plate resistance (R_P) of the two tubes involved. Most handbooks have this figure. Next, check the circuit diagram and get the value of the plate load resistor which you are using. This is the resistor which connects directly to the plate at one end and is bypassed to ground (and connects to B plus) at the other end. Next, get the value of grid resistor on the tube which follows the tube whose value of R_P you just looked up. Now, calculate the effective parallel resistance of these three factors, that is, of R_P, the plate resistance, of R_L, the plate load resistance, and R_G, the grid resistance, by the formula:

$$\frac{1}{R_T} = \frac{1}{R_P} + \frac{1}{R_L} + \frac{1}{R_G}$$

For example, assume that a 6J5 tube uses a plate load resistor of 50,000 ohms. The plate resistance of a 6J5 is approximately 7000 ohms. Assume also that the grid resistance of the next stage is 250,000 ohms. The effective resistance of these three in parallel is 5990 ohms. Call this R_T for the 6J5 stage. Incidentally, the R_P for triodes is low, as shown above. For pentodes, R_P will be very high.

The proper value of shunt condenser to connect from plate to ground is one whose capacitive reactance, at 3500 cycles, is equal to R_T. Stated again simply, the value in micro-farads is:

$$\frac{1{,}000{,}000}{(22{,}000)\ (R_T)}$$

This assumes that the high frequency point selected was 3500 cycles. The figure of 22,000 is 3500 times 2 times π. As an example, if R_T is 5990 ohms, then the plate to ground condenser calculates out to be 0.0076 mf so use a 0.0075 mf condenser. Connect it to the plate of the tube and to a convenient ground point. Make this calculation for both stages. This takes care of the higher frequency audio tones.

Let us now examine the change we have brought about in the speech amplifier and also examine what we have gained from this change. To do this, we shall have to assume that the response of the speech amplifier, before the change, was fairly uniform from 150 to 6000 cycles. This is the sort of response which might be expected in a speech amplifier following general circuit practice. In addition, the response was probably only five or six db down at 100 and 10,000 cycles.

When you used your speech amplifier, before the change, you were modulating your carrier with all the complex audio tones that existed in the microphone output, over the 100 to 10,000 cycle range. Your sideband power, which is all that the other ham is using to hear your signal, was therefore spread over a wide frequency range. It so happens that it takes a fair amount of modulator power to transmit the lower and higher frequency audio components which are not necessary for intelligibility.

By making the change in your speech amplifier, you now still have the same power in your sidebands, assuming that the percentage of modulation is the same, but you now have a great deal more power available to transmit the range of frequencies that really count, those between 300 and 3500 cycles. Effectively, therefore, you have a "louder" signal, because you have increased power at the audio frequencies to which the other ham listens. In round numbers, the increase in signal strength is about 6 db, which is the same as a four to one increase in carrier power, or the same as putting up an antenna with a 6 db gain over the one you were using.

To get an idea of the response curve which is obtainable, let us look at a speech amplifier which uses, for example, a 6SL7 dual triode for the first two stages, driving a third stage which has a 250,000 ohm grid leak. Assume that the aforementioned changes have been made. Now let us apply a pure tone at 1000 cycles, the midband frequency, and measure the output of the speech amplifier. Next, apply a pure tone of 300 cycles. The output will be down 6 db, or four to one in power. The same thing is true for a 3500 cycle tone. A pure tone at 150 cycles (and at 7000 cycles) will be down 14 db, or twenty-five to one in power.

Thus, while the curve obtained is not of the sharp cutoff variety, it will give essentially the same results, and will certainly sound the same to the ear. Further, it was obtained at practically no cost.

—Lighthouse Larry.

RESTRICTED RANGE SPEECH AMPLIFIER

Audio Amplifier Designed Expressly for Speech Work

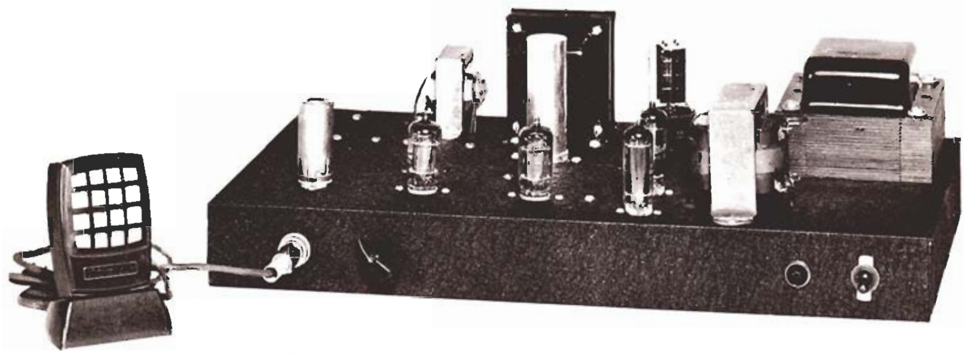

Fig. 1. Front View of the Restricted-range Speech Amplifier

From September-October, 1949

FEATURES—

- "Speech" range from 500–2500 cycles
- Minimum distortion
- Five miniature tubes—one rectifier
- Power output of 10 watts

GENERAL CONSIDERATIONS

A speech amplifier for amateur radio service has the job of amplifying the human voice until the complex waveform which forms the human voice has sufficient power to drive the modulator tubes. The amplifier's job, then, is relatively simple. However, the frequency characteristic of the audio amplifier—that is, the amount of amplification which will be obtained at various audio frequencies—determines to a large degree the type of radio-frequency signal which is put on the air.

For example, if you are using a speech amplifier capable of amplifying frequencies beyond ten thousand cycles, and your voice (or extraneous background noise) contains energy at this frequency, then the radio-frequency signal from your transmitter will extend out at least ten thousand cycles—10 kilocycles—on each side of your transmitted radio frequency. Stated another way, your signal has a minimum width of 20 kilocycles. Broad? Quite broad. Even aside from the fact that you have a broad signal, there is little point in transmitting a high fidelity signal. Primarily this is because the average communication receiver does not have an audio system capable of reproducing these high frequencies. In addition, a highly selective receiver will further restrict the audio frequency characteristic.

If you use another amplifier which has practically no gain at 10,000 cycles, but which drops rapidly in gain past 5000 cycles, then this same voice, using this amplifier, will modulate the radio-frequency carrier so that energy exists out 5 kilocycles each side of the center frequency. This gives a signal with a width of 10 kilocycles. By using this second amplifier, have you lost naturalness, does your voice sound exactly the same to the amateur receiving it over the air as it would if he heard you in person? No. Can you be understood? Yes.

How far can this process be carried? How much can we restrict the bandwidth of the speech amplifier, and still have voice modulation which is adequate for communication purposes? While it is impossible to give an answer to this question which will satisfy everyone, most engineers agree that a bandwidth, for understandable speech, of 500 to 2500 cycles is adequate. This is not as narrow a band as might be imagined. For example, the major radio networks send their programs to their member stations on telephone lines. The best of these lines have a cutoff frequency of approximately 5000 cycles. Certainly

ELECTRICAL CIRCUIT

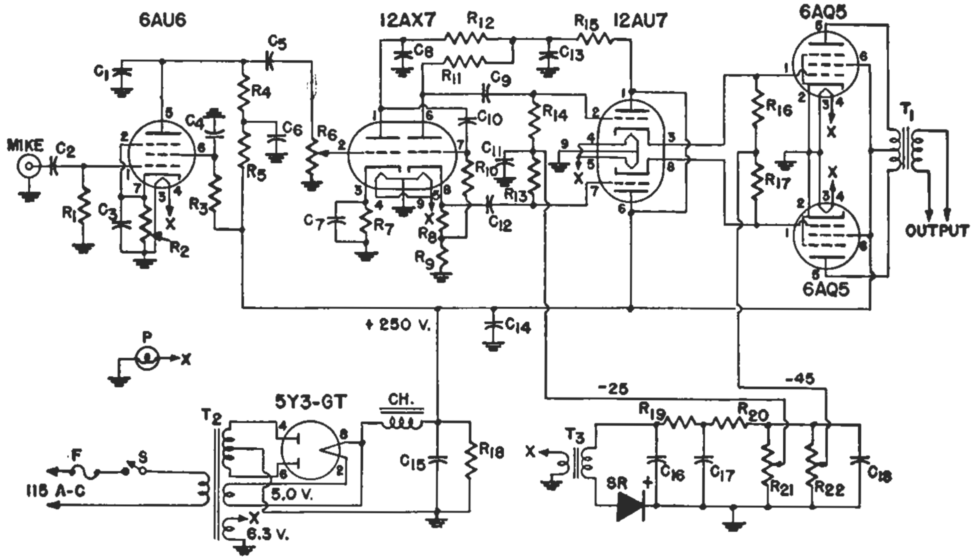

Fig. 2. Circuit Diagram of the Restricted-range Speech Amplifier

CIRCUIT CONSTANTS

C_1 . . . . 1590 mmf mica (see text)
C_2 . . . . 0.005 mf 600 volt paper
C_3, C_7 . . . . 50 mf 50 volt electrolytic
C_4, C_6, C_{13}, C_{14} . . . . 8 mf 450 volt electrolytic
C_5 . . . . 1275 mmf mica (see text)
C_8 . . . . 1380 mmf mica (see text)
C_9, C_{12} . . . . 0.01 mf 600 volt paper
C_{10} . . . . 580 mmf mica (see text)
C_{11} . . . . 0.1 mf 600 volt paper
C_{15} . . . . 40 mf 450 volt electrolytic
C_{16}, C_{17}, C_{18} . . . . 40 mf 150 volt electrolytic
CH. . . . . 8 henry smoothing choke, 150 mils
F . . . . 2 ampere fuse
P . . . . 6.3 volt pilot light
R_1 . . . . 1 megohm, ½ watt
R_2 . . . . 5500 ohm, 1 watt
R_3 . . . . 0.47 megohm, 1 watt
R_4 . . . . 47,000 ohm, ½ watt
R_5 . . . . 10,000 ohm, 1 watt
R_6 . . . . 0.25 megohm potentiometer
R_7 . . . . 3500 ohm, 1 watt
R_8 . . . . 2500 ohm, ½ watt
R_9 . . . . 17,500 ohm, ½ watt
R_{10} . . . . 0.1 megohm, 1 watt
R_{11} . . . . 20,000 ohm, 1 watt
R_{12} . . . . 0.33 megohm, 1 watt
R_{13}, R_{14} . . . . 0.5 megohm, ½ watt
R_{15} . . . . 6600 ohm, 6 watt (three 20,000 ohm, 2 watt resistors in parallel)
R_{16}, R_{17} . . . . 6000 ohm, 1 watt
R_{18} . . . . 50,000 ohm, 10 watt
R_{19}, R_{20} . . . . 1000 ohm, 2 watt
R_{21}, R_{22} . . . . 10,000 ohm, 10 watt semi-adjustable
S . . . . SPST toggle switch
SR . . . . Selenium rectifier (G-E 6RS5GH2)
T_1 . . . . Output transformer (see text)
T_2 . . . . Power transformer, 350–0–350 at 150 mils, 6.3 volts at 4 amperes, 5.0 volts at 3 amperes
T_3 . . . . . . Filament transformer, 6.3 volts at 1 ampere

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we do not think of network broadcasts as having "poor quality," and yet 5000 cycles (approximately) is the highest audio tone which will be heard when listening to network programs.

The primary advantage in using a speech amplifier which has a restricted high-frequency response is that the radio-frequency signal resulting will occupy less space in the spectrum. Recent FCC amateur proposals which refer to the bandwidth of radio-frequency signals (on which no action has been taken at the time of this writing) can be complied with most easily by sufficient reduction in the response of the speech amplifier at the higher audio frequencies. This is because the radio-frequency bandwidth of a properly operated transmitter is dependent only upon the range of the audio frequencies used to modulate the transmitter. This assumes that the transmitter is free of parasitics, is operating on only one frequency, and the modulation applied is within the modulation capability of the modulated stage, to cite a few of the effects which may give a broad signal, even though the modulating frequencies are within the proper range.

(However, in the case of NBFM, the use of a restricted-range speech amplifier will not assure that the radio-frequency signal does not occupy too much space. If the frequency swing caused by the modulation is excessive then the radio-frequency signal will be unnecessarily broad.)

Thus far we have discussed primarily the higher-frequency audio tones. However, it is also desirable to eliminate, or attenuate, the low frequency audio response of the speech amplifier. Elimination of all response below, say, 500 cycles, would have no effect on the width of our radio-frequency signal, but it would give us the effect of a stronger signal. It is difficult to put an actual number on the gain which could be achieved, but with relatively simple attenuation means used in the speech amplifier a gain of 5 to 6 decibels would be possible. This is the sort of gain which can be expected from a good two-element parasitically excited beam, or by increasing your power by a factor of four.

The energy output of the male voice is concentrated at the lower frequency end of the audio frequency spectrum. Unfortunately these low frequency components of the male voice contribute little to the intelligence in speech. However, being of high amplitude, a great deal of modulator power and modulation capability is required to transmit them. Obviously we can increase the effective transmitted power by reducing the number of low frequency components in the system. Paradoxically a system with restricted high-frequency response, such as discussed previously, sounds more natural if the low frequency components are attenuated in a balanced manner.

There are many ways to accomplish the desired attenuation of the lower and higher frequency portions of the audio-frequency spectrum. All of these methods use an audio network, either simple or complex, which will attenuate certain frequencies either more or less than other frequencies. The amount of attenuation achieved will depend upon the type of network used.

Referring again to the April 27 FCC proposals no statement has been made public, at the time of this writing, as to the amount of attenuation that the FCC feel is adequate. The attenuation achieved in the amplifier about to be described is shown in Fig. 3. This attenuation averages 12 db per octave. Stated another way, the power is down by a factor of sixteen for each octave considered. For example, the power output of the speech amplifier at 10,000 cycles is one-sixteenth of the power output at 5000 cycles.

Referring again to Fig. 3, the calculated operating range of the speech amplifier is from 500 to 2500 cycles. Note that the curve is not flat over this portion, but that the 500 and 2500 cycle points are approximately 6 db down from the midpoint frequency, which is approximately 1000 cycles. For the first octave below 500 cycles and the first octave above 2500 cycles, the attenuation has not yet reached a slope of 12 db per octave. However, for further octave jumps the attentuation will be quite close to 12 db per octave, so that the 125 and 10,000 cycle points will be down by 26 db and the 62 and 20,000 cycle points down by 38 db.

Note this 62 cycle point. The attenuation at this point is theoretically 38 db or, as actually measured in the speech amplifier, 35 db. This means that the power output at 62 cycles will be only one four-thousandth of the power output at 1000 cycles. This means that normal precautions regarding sixty cycle hum need not be taken. As a result, the filament wires in this speech amplifier were neither paired and twisted nor carefully handled. One side of each filament connection was grounded and the other lead run as a single wire. While this may not seem startling, those of you who have had trouble with hum in high-gain amplifiers will appreciate this statement.

The design procedure used in this speech amplifier is identical to that discussed in the "Technical Tidbits" section of the July-August, 1949 G-E Ham News. Readers are referred to this article for the back ground work on the present design. Suffice it to say that C_5 and C_{10} (see Fig. 2) have the job of attenuating the low frequency end of the audio spectrum, and C_1 and C_8 handle the attenuation of the higher audio frequencies. In other words, the entire job is handled by the proper choice of four condensers, two of which would normally be employed in the amplifier even if a restricted bandwidth were not desired.

ELECTRICAL DETAILS

Referring to the circuit diagram, Fig. 2, the tube functions are as follows. The 6AU6 serves as a pentode voltage amplifier, giving a mid-band gain of well over 100. The first section of the high-mu double-triode 12AX7 serves as the second voltage amplifier, and gives a gain of approximately 50. The second section of the same tube acts as a phase inverter. The 12-AU7 tube is a push-pull cathode follower which acts as a low-impedance driving source for the push-pull 6AQ5 output tubes. It is absolutely essential that distortion be held to as low a value as possible if full advantage is made of the restricted bandwidth of this speech amplifier. This is because distortion will cause the radio-frequency signal to become broad, and this is one of the effects that we wished to overcome by restricting the audio bandwidth.

One of the major causes of distortion in the audio systems of amateur transmitters is the use of driver stages with too high an internal impedance to properly drive class AB_1 or Class B stages. Distortion results, in this case, because of poor regulation in the driving voltage when the driver is called upon to supply the grid current drawn during voltage peaks. The 12AU7 cathode follower tube acts as a *low-impedance* driver. This permits more power output from the 6AQ5 tubes with less distortion than would be possible if the 6AQ5 tubes were driven directly from the phase inverter.

Essentially the 6AQ5 tubes are operated as class AB_1 amplifiers. Normally this means that no precautions need be taken with the driver stage to ensure minimum distortion provided that the grids are

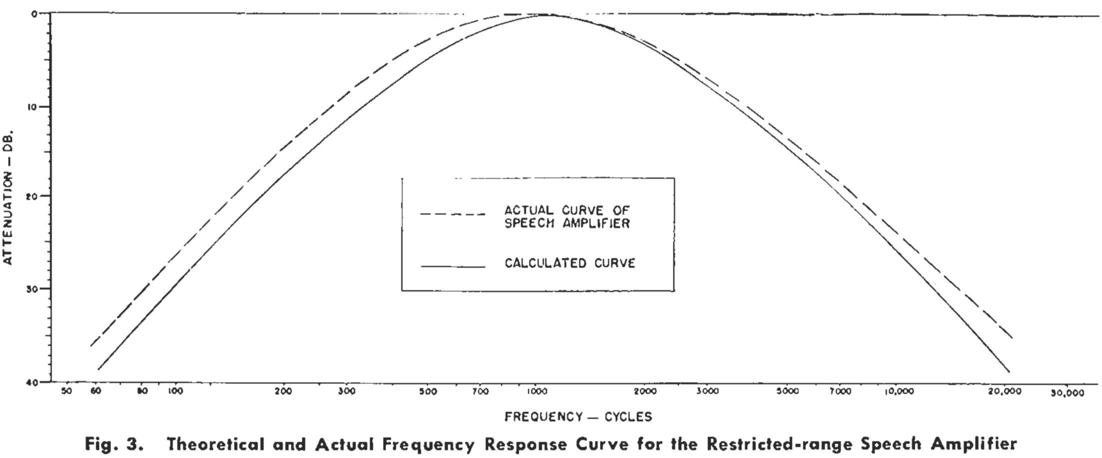

Fig. 3. Theoretical and Actual Frequency Response Curve for the Restricted-range Speech Amplifier

never driven positive. This condition is difficult to achieve unless the average level is kept quite low. By using a driver which presents a low source impedance, which the 12AU7 accomplishes, the average level may be pushed up quite high and the 6AQ5 tubes driven all the way up to the grid bias point. Even if an occasional voice peak causes this voltage point to be exceeded, no distortion will occur due to "folding-up" of the driver stage. The net result is high output, minimum distortion, and a "narrow" radio-frequency signal.

Condensers C_1, C_5, C_8 and C_{10} (the frequency controlling condensers) are listed in "Circuit Constants" with values which are not stock values. The values shown are those which calculation indicate to be correct. Try to obtain condensers moderately close to these values. It is not wise to trust the values marked on condensers, and it is recommended that a capacitance bridge be borrowed to check through your stock of mica condensers. It may be easier to parallel condensers in order to get the proper value. For example, C_1 could easily be made up with a 1000 mmf and a 600 mmf condenser in parallel.

One further point might be made, with reference again to the circuit diagram. Fixed bias is supplied to two stages, the 12AU7 stage and the 6AQ5 stage. The bias supply is unusual in one respect. Cathode current for the 12AU7 stage must flow through R_{22}. The total current for both 12AU7 sections is approximately 10 mils. In other words, this bias supply must be capable of supplying a voltage and a current, instead of just a voltage as in the usual case. If the circuit diagram is followed no difficulty will be encountered. However, if you attempt to use another source of bias, make certain that it can supply the required current.

MECHANICAL DETAILS

The amplifier was constructed on a 17 by 10 by 2 inch chassis. However, inasmuch as practically any layout scheme will work, the prospective builder can use any convenient size chassis and change the layout to suit. The entire speech amplifier and power supply could fit easily in a chassis of half the size of the one just mentioned.

The placement of parts can best be seen in Fig. 1. The tubes are, from left to right, 6AU6, 12AX7, 12AU7, 6AQ5's, and the 5Y3-GT rectifier tube in the rear. Note that the 6AU6 uses a shield. On the rear of the chassis, at the left, is the bias transformer, T_3, with the choke and C_{15} to the right. The power transformer occupies the rear corner and the output transformer is directly ahead of it.

Only two controls are employed—the on-off switch and the gain control. The microphone input jack and the pilot light are mounted on the front of the chassis, and the fuse on the rear of the chassis.

The underchassis view of the amplifier, Fig. 4, indicates the placement of the remainder of the components. No shielded wire was used, mainly because all leads to the first two stages were short. If the layout is altered from that indicated, it might be advisable to shield any long leads in the first two or three stages.

OPERATING ADJUSTMENTS

Once the amplifier has been completed, and it has been established that voltage can be applied without anything smoking, the 12AU7 bias voltage and cathode-return voltage should be adjusted. R_{21} should be adjusted so that the bias, as read from the arm of R_{21} to ground, is 25 volts. Adjust R_{22} until the voltage from the arm of R_{22} to ground is 45 volts. Next check the bias on the 6AQ5 tubes by reading the voltage from pin 1 of either tube to ground. This voltage should be 15 volts. If this is not true, change the tap on R_{21} slightly until the 6AQ5 bias (pin 1 to ground) reads 15 volts. The 45 volt cathode-return voltage should remain unchanged during this adjustment. It will not be necessary to have any input signal to the speech amplifier during the foregoing tests.

The last check to be made, assuming that the am-

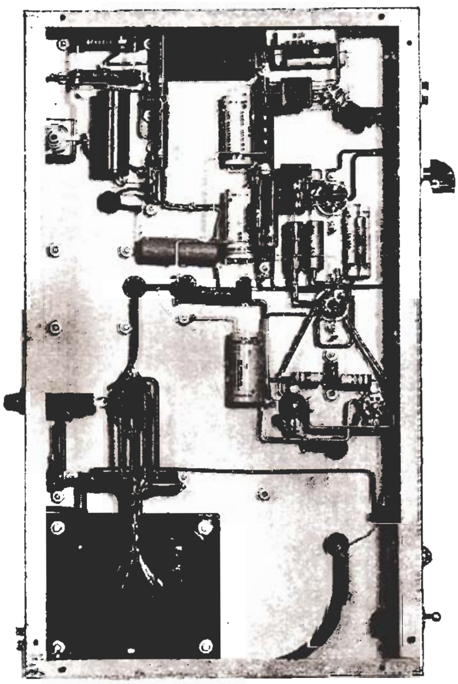

Fig. 4. Under-chassis View of the Restricted-range Speech Amplifier

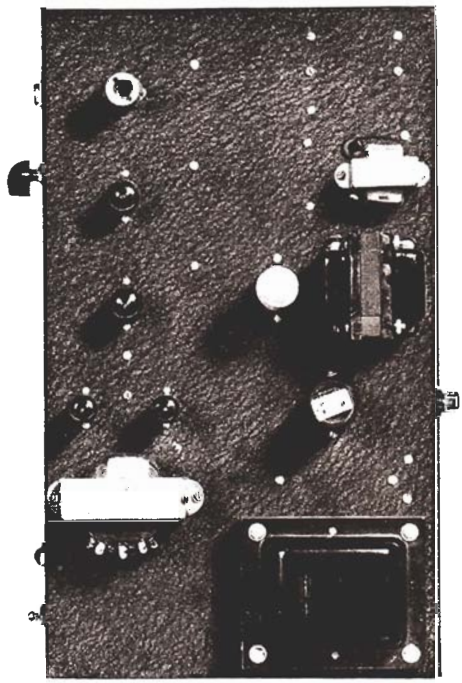

Fig. 5. Top View of the Restricted-range Speech Amplifier

plifier has been correctly wired and is operable, is to match the 6AQ5 tubes to their load. The amount of power that these tubes can deliver will depend to a great degree upon the output transformer. The selection of an output transformer will be governed primarily by the proposed application. It is recommended, however, that a transformer with a number of various impedances be used, so that minor changes in matching may be made.

It is further recommended that a transformer be purchased which has a generous power rating. For example, a 10 watt transformer will serve, but a 20 watt output transformer will permit more output to be achieved without distortion. The amplifier pictured uses a 10 watt output transformer. The highest output power which could be achieved without discernible distortion on an oscilloscope was 7.2 watts (measured output from the transformer). A second amplifier with an 18 watt output transformer permitted an output, under the same conditions, of 11.2 watts. In both cases the transformer was matched to the output load impedance, which took the form of a resistor.

Therefore, procure a transformer which is capable of matching from an approximate impedance of 10,000 ohms (the plate-to-plate effective load resistance of the 6AQ5 tubes) to whatever class B grids you wish to drive. Or, you may wish to match 10,000 ohms to a 500 ohm line. In the latter case another transformer is required to match from the 500 ohm line to the modulator grids. If this system is used, approximately twice as much power is lost between the driver plates and the modulator grids as compared to the case where only one transformer is used. You may expect to get losses up to 3 db in each transformer. Three db is two-to-one in power.

Once the transformer is procured and the amplifier tested while driving the required load, it may be advisable to make small changes in the impedance ratio between driver and modulator to ensure that you have an impedance match which will give maximum power transfer with minimum distortion.

LOGARITHMIC COMPRESSOR

Aids in Preventing Overmodulation While Increasing Signal Effectiveness

From May-June, 1950

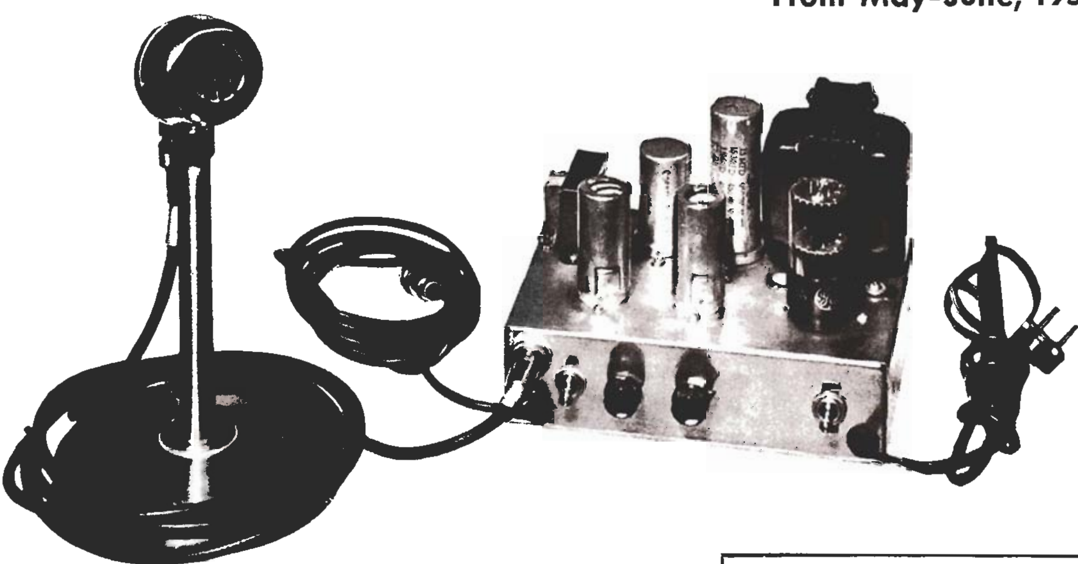

Fig. 1. The Logarithmic Compressor ready to plug into your present microphone jack. Controls are, left to right, in-out switch, compression control, output control and a-c on-off switch.

FEATURES—

Provides 10 db increased effectiveness

Uses self-contained speech-range filter

Three tubes, including rectifier

Small size—space saving

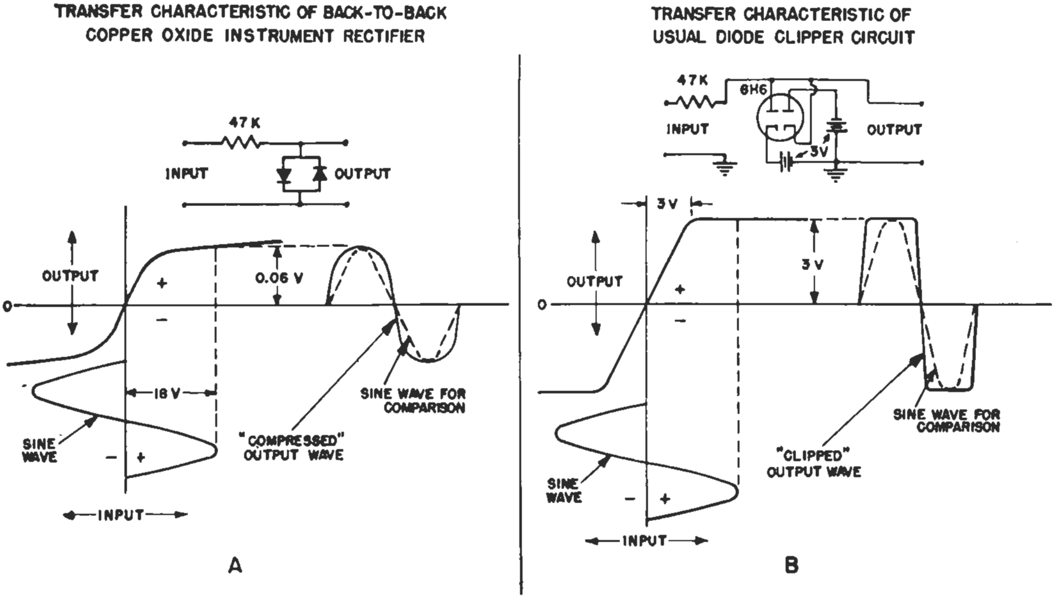

Fig. 2. A comparison between the output waveform of a Logarithmic Compressor and a diode clipper.

Every phone man, at some time in his QRM-ridden life, has wished that he had a small switch available which would permit him magically to increase his power tenfold. This would be Utopia—from one kilowatt to ten kilowatts by pressing a button.

This button is now available, and it is mounted on the front of the Logarithmic Compressor. This unit will give an effective signal gain which is adjustable from a few db up to as much as ten db (ten to one in power).

The Logarithmic Compressor is an audio amplifier device which is inserted between your microphone and your present speech amplifier. Its function is to push up the average modulation level, with the result that high percentage modulation is assured at all times, regardless of the sound level reaching the microphone.

COMPRESSION VS. CLIPPING

Those familiar with clippers or clipping circuits can see that the Logarithmic Compressor is intended to do the same sort of job as a clipper. There is, however, an important difference between logarithmic compression as used in the Logarithmic Compressor and clipping.

Fig. 2 compares the characteristics of the two different systems. In either case the input wave suffers distortion, but the distortion caused by the clipping action of the ordinary diode type clipper (Fig. 2B) is worse for a given amount of signal compression than that caused by the logarithmic compression of a copper-oxide instrument rectifier (Fig. 2A).

Distortion present in either circuit will add "harshness" to speech signals and without further treatment would result in excessively broad signals. Therefore, any distorting type circuit should be followed by a suitable filter to prevent the high frequency products produced by this distortion from reaching the modulated stage. With such a filter much of the "harshness" will still be present but the radio-frequency signal need not be broad. The harshness results from cross modulation (distortion) products that lie within the pass band of the filter.

The advantage of the logarithmic compression system is that the distortion is less severe (for a given amount of compression) than the clipper type, and this makes possible the use of a vastly simpler filter arrangement. Three "stages" of R-C type filtering used in the Logarithmic Compressor are as effective as more elaborate sharp-cutoff types of L-C filter virtually necessary with the clipper type of circuit.

Further, the transient response of the R-C type filter is such that no overshoot of signal peaks can occur. This is not the case with sharp-cutoff L-C filters. This means that the logarithmic compressor circuit with a properly designed R-C filter is superior to the ordinary clipper circuit followed by a sharp L-C filter. Repeated tests confirm this statement.

CIRCUIT DETAILS

With reference to Fig. 3 it will be seen that the first 12AT7 acts as a two stage audio amplifier to bring the signal from the microphone to a sufficient level so that the compression circuit itself operates at the proper level. Resistor R_1 in the first stage has been added as a precaution against radio-frequency feedback.

Special care has been taken to attenuate low audio frequencies prior to compression. Doing this gives a well balanced speech response as well as minimizing much of the distortion caused by cross-modulation between the low speech frequencies and the *intelligence-bearing* high speech frequencies. The values of condensers C_2, C_3 and C_4 are chosen to attenuate the low frequencies adequately before speech compression. Condensers C_7 and C_9 serve the same purpose after compression has taken place.

Resistor R_4, by varying the signal input to the second section of the first 12AT7, enables control of the amount of compression.

The audio transformer, T_1, is necessary because the limiting circuit must be fed by a low-impedance, low-resistance source. Using the center-tap on this transformer accomplishes this function.

The actual limiting or compression circuit consists

ELECTRICAL CIRCUIT

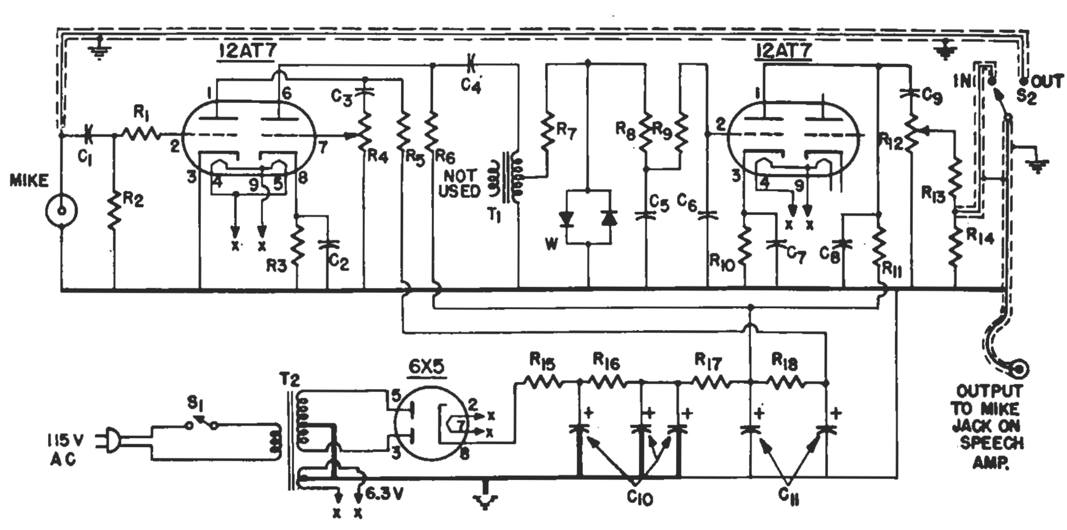

Fig. 3. Circuit diagram of the Logarithmic Compressor.

CIRCUIT CONSTANTS

(All resistors and capacitors ±20% tolerance unless specified otherwise)

C_1, C_9 .... 0.01 mf 400 volt paper or mica
C_2, C_7 .... 1.0 mf paper (see text)
C_3 .... 1000 mmf mica
C_4 .... 0.05 mf 400 volt paper
C_5 .... *1000 mmf ±10% mica*
C_6 .... *100 mmf ±10% mica*
C_8 .... *0.005 mf ±10% mica*
C_{10} .... 15-15-15 mf 450 volt electrolytic (see text)
C_{11} .... 10-10 mf 450 volt electrolytic
R_1, R_{14} .... 10,000 ohm, ½ watt
R_2 .... 10 megohm, ½ watt
R_3, R_{10} .... 470 ohm, ½ watt
R_4 .... 0.5 megohm potentiometer
R_5 .... 0.1 megohm, 1 watt
R_6, R_{11} .... 47,000 ohm, 1 watt
R_7 .... 47,000 ohm, ½ watt
R_8 .... *56,000 ohm, ±10%, ½ watt*
R_9 .... *0.56 megohm, ±10%, ½ watt*
R_{12} .... 0.1 megohm potentiometer
R_{13} .... 0.47 megohm, ½ watt
R_{15} .... 470 ohm, 2 watt
R_{16} .... 2200 ohm, 2 watt
R_{17} .... 1000 ohm, 1 watt
R_{18} .... 4700 ohm, 1 watt
S_1 .... SPST toggle switch
S_2 .... SPDT toggle switch
T_1 .... Push-pull plates to voice coil audio transformer, UTC R-38A (see text)
T_2 .... Power transformer, 300-0-300 volts at 50 mils, 6.3 volts at 2 amperes UTC R-54
W .... Copper-oxide instrument rectifier (see text)

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only of R_7 and W, the latter being two sections of a copper-oxide instrument rectifier. Resistors R_8 and R_9, together with condensers C_5 and C_6 act as a two-section R-C filter. The output of this filter feeds the second 12AT7 directly. Resistor R_{12} acts as an output control so that the output level from the speech compressor may be made to match the output level of the microphone. Thus when the speech compressor is switched out of the circuit no other adjustment need be made.

The output tube is required for two reasons. It is necessary to present the proper load to the two R-C filters and, secondly, to permit a third R-C stage to be utilized. Inasmuch as the second section of the 12AT7 tube is not used this may seem like wasting part of the tube, but the use of a high-mu triode was dictated and the 12AT7 fills this requirement nicely. Note that the heater of the unused section need not be energized. Many uses for this extra tube section will undoubtedly suggest themselves.

The in-out switch, S_2, allows the unit to be switched in and out of the circuit easily. Note that shielded wire is specified for the connections to this switch. The output itself is carried by a shielded lead which plugs into the mike jack of any speech amplifier designed to handle a high impedance dynamic or crystal microphone.

The power supply is conventional in all respects. Because of the low current drain on the power supply a resistor-capacitor filter is employed. Resistor R_{18} and condenser C_{11} provide decoupling and additional filtering for the first 12AT7 section plate voltage.

The connections indicated by the heavy black lines in the power supply section should all be made to one ground point. This will prevent the chassis from carrying the circulating capacitor current and help to keep the unit hum-free.

CONSTRUCTIONAL DETAILS

As may be seen from the photographs, the entire unit, including power supply, is mounted on a 5 by 7 by 2 inch chassis. While the layout is not critical, it is advisable to keep the power supply portion of the circuit as far away from the rest of the circuit as possible. The layout shown is quite satisfactory.

With reference to Fig. 1, the front panel layout, from left to right, is: mike jack, output lead, in-out

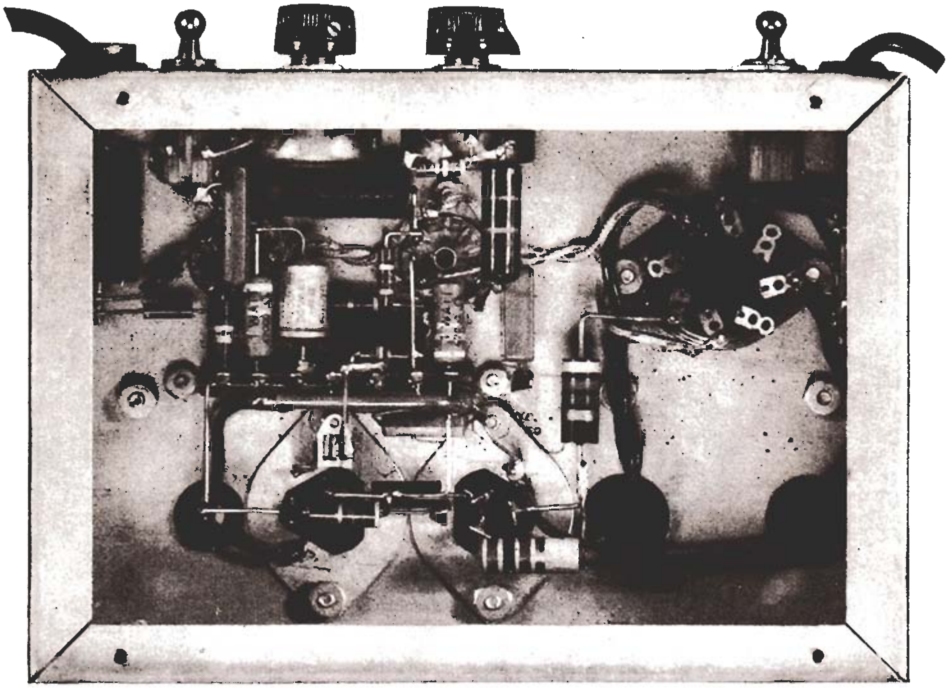


Fig. 4. Under-chassis view of the Logarithmic Compressor. There is ample room for all component parts.

switch, compression control, output level control, a-c on-off switch and a-c cord. The tubes are, left to right, input 12AT7, output 12AT7 and 6X5 rectifier. Note that the two 12AT7 tubes are shielded.

Fig. 4 gives the details of the wiring. Nothing here is critical if normal wiring procedure is followed. Note that R_1 is placed as close to the grid pin as possible.

The wiring can be made simpler if the unused leads from the power transformer are pulled inside the transformer case and securely taped to avoid shorts. This was done with the 2.5 volt and the 5.0 volt windings.

The unit pictured uses a bottom cover plate for the chassis. This is recommended to avoid r-f feedback. Any sort of thin metal will serve for this purpose, if your chassis comes without a bottom plate.

COMPONENT PARTS

While no extremely critical values are required, it is recommended that the specified values be used in all cases. For example, C_2 and C_7 are specified as 1.0 mf condensers. If lower values were to be used, the frequency response would suffer, and if higher values were used, the result would be insufficient low-frequency attenuation.

Condenser C_{10} is about the only component which could be changed. Here a 20-20 mf condenser could be used, with one of the 20 mf section on either side of R_{16}.

Almost any sort of push-pull plates to voice coil transformer will serve as T_1. Wattage rating of this transformer is not important.

If possible, linear taper potentiometers should be used at R_4 and R_{12}. This sort of taper will give a smoother action than other types of taper.

Care must be taken in purchasing the limiter rectifier, W, because instrument rectifiers come in several different styles. Basically, of course, they are used to make a-c meters out of d-c meters. However, they can be purchased as half-wave units, doubler units, full-wave units and bridge units.

Two separate half-wave units, connected as shown, will work, and the bridge-rectifier style will work if the proper leads are used. The "full-wave" unit will not serve because the two diode sections are connected improperly. In the doubler type rectifier the two diode units are connected as shown in Figs. 2 and 3 and therefore this type of instrument rectifier would be the best to use.

COMPRESSION ADJUSTMENT

The adjustment of the Logarithmic Compressor is done very easily. Plug in a mike and place the in-out switch, S_2, in the "out" position so that the microphone is connected directly to your speech amplifier, then follow these three steps:

1. Adjust the audio gain control on the transmitter for normal modulation as seen on an oscilloscope (the best method) or some other instrument worthy of trust.

2. Put the output control on the unit to zero and set the compression control so that it is about half open. Switch the compressor to "in" and advance the output control while speaking into the microphone until the ***peak*** modulation is the same as in step 1. While an oscilloscope is not absolutely necessary in order to make this adjustment, it is strongly recommended.

3. Adjust the compression control so that the average plate current in the modulator stage on a sustained "00000——0" is, say, not over twice that obtained with the compressor out. Then try compressor "in" and "out" on a few QSO's to find the best operating point of the compression control for the microphone you are using and the receiving conditions prevalent at the other fellow's QTH.

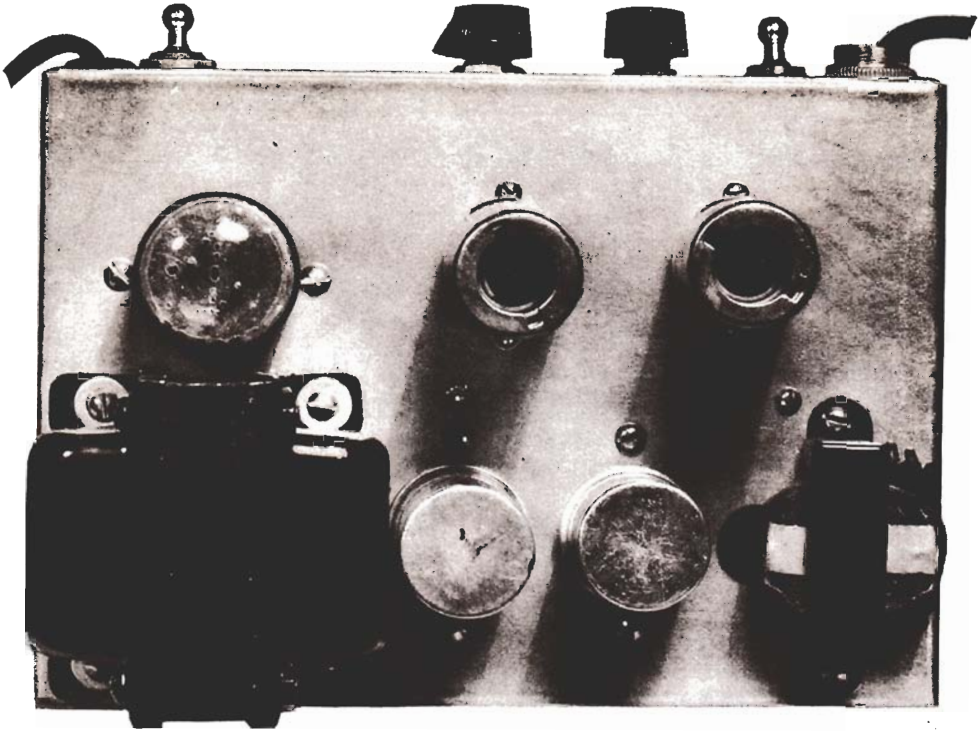


Fig. 5. Top view of the Logarithmic Compressor. The audio section is on the right and the power supply section is on the left.

USE OF THE COMPRESSOR

With the Logarithmic Compressor in use the modulator tubes are required to handle much more average power than usual. In fact, it is possible that your modulator stage will not be capable of handling the extra average power required. Careful checking with an oscilloscope will determine if this is the case.

As a general rule, if your modulator can handle a sine wave signal at 100% modulation, then the average power capability of your modulator is adequate for use with the Logarithmic Compressor. (After all, this ten db gain has to come from some place!) This means that, for a kilowatt rig, your modulator should be capable of continuous operation at 500 watts output at 1000 cycles. For lower powers the same ratio holds.

In operation the compressor must be used with judgment—good judgment that is. Too much compression may make an otherwise acceptable signal almost intolerable. With a judicious amount of compression one can expect to add from 6 db (4 to 1 in power) to 10 db (10 to 1 in power) in the effectiveness of his signal *provided* conditions at the receiving point are such that understandability without the compressor is impaired by QRM or high background noise.

RESULTS WITH THE COMPRESSOR

In many months of test at W2KUJ the following information has been obtained. Nearby stations, or stations not experiencing QRM, prefer that the compressor *not be* used. Stations receiving a weak signal or listening through severe QRM prefer that the compressor *be* used.

Reports from the latter stations range from eight to ten db jump in effective signal strength when the compressor is switched in. Reports from nearby stations are that the signal is *louder,* but somewhat less readable with the compressor in use than without it.

In no case has a report been given that the signal was broader when the compressor was used, even when this question was asked of nearby stations.

Tests made at W2RYT's shack indicate that different microphones give somewhat different results when used with the compressor. For example, an Electro-Voice Model 605 dynamic mike (pictured in Fig. 1) and an Electro-Voice Model 915 crystal mike seemed to have identical speech characteristics (although the dynamic mike had less output) when used without the compressor.

When used with the compressor, the dynamic mike was found to have a speech quality which was less harsh than that of the crystal mike. Further, it was found advisable to advance the compression control with the dynamic mike.

The foregoing is not intended as a recommendation for dynamic mikes, nor is it intended as an authoritative comparison between two Electro-Voice microphones. The comparison has been made to emphasize the importance of testing your compressor carefully with each microphone you may use with it.

In summary, one can expect to boost the effectiveness of his signal when it is needed most by use of the compressor (it frequently means the difference between making a contact or not) with some decrease in ease of reading the signal where the compressor is not needed.

Bear in mind that the compressor *can be* misused (to your disadvantage). Seek honestly to find the operating points which best exploit its use. In many cases it is best to *not* use the compressor. But in those cases where it is needed, the Logarithmic Compressor can really do a job for you.

Added Information for Logarithmic Compressor

The comments below answer many of the questions regarding the copper oxide rectifiers and other components in this unit, plus applying the circuit to existing audio equipment for single sideband and other amateur transmitters.

1. COMPONENTS PARTS - RECTIFIERS- Suitable copper oxide instrument rectifiers (W in schematic diagram) are made by several manufacturers. The following list includes rectifiers shown in the catalogs of several mail order electronic components suppliers (Allied Radio, Radio Shack, etc.):

Schauer Mfg. Corp. 4513 Alpine Ave. Cincinnati, Ohio	1-doubler type A2MC, or, 2- half wave types A1P
Conant Laboratories 6500 "Q" st. Lincoln, Nebraska	1- doubler type 160-BHS, or 2-half wave types160B
Bradley Laboratories 168 Columbus Ave. New Haven 11, Conn.	1- doubler type CX2E series, or 2- half wave types CX2E series

2. COMPONENT PARTS - TRANSFORMER T_1 -- A UTC type R-38A universal output transformer (push-pull plates to voice coil) was recommended for this circuit. Any similar transformer made by other firms may be used instead. Similar transformers are: Stancor A-3856, Thordarson 24S60, Merit A-3936 or A2938, Halldorson Z1404, Triad S-15X, and Freed RGA-11. Certain center-tapped audio chokes may be used, if available. Try the choke you happen to have and check the operation of the compressor with an oscilloscope. A waveform similar to that shown in Fig. 2A in the May-June, 1950 issue should be obtained for best results.

3. SUBSTITUTING TUBE CATHODE FOLLOWER FOR T_1 -- Although some experimenters have reported moderate success with a cathode follower tube instead of T_1 (the unused half of one 12AT7 in the original circuit), best results will be obtained with the original circuit components. These parts and values were determined by many hours of laboratory tests. The transformer, T_1, provides a low-resistance path for the non-linear signal applied to the rectifiers. A positive bias (from the current flow through the cathode resistor) will be applied to the copper oxide rectifiers when they are driven from a cathode follower, unless a large coupling capacitor (about 20 mfd.) is placed between the tube cathode and the rectifiers.

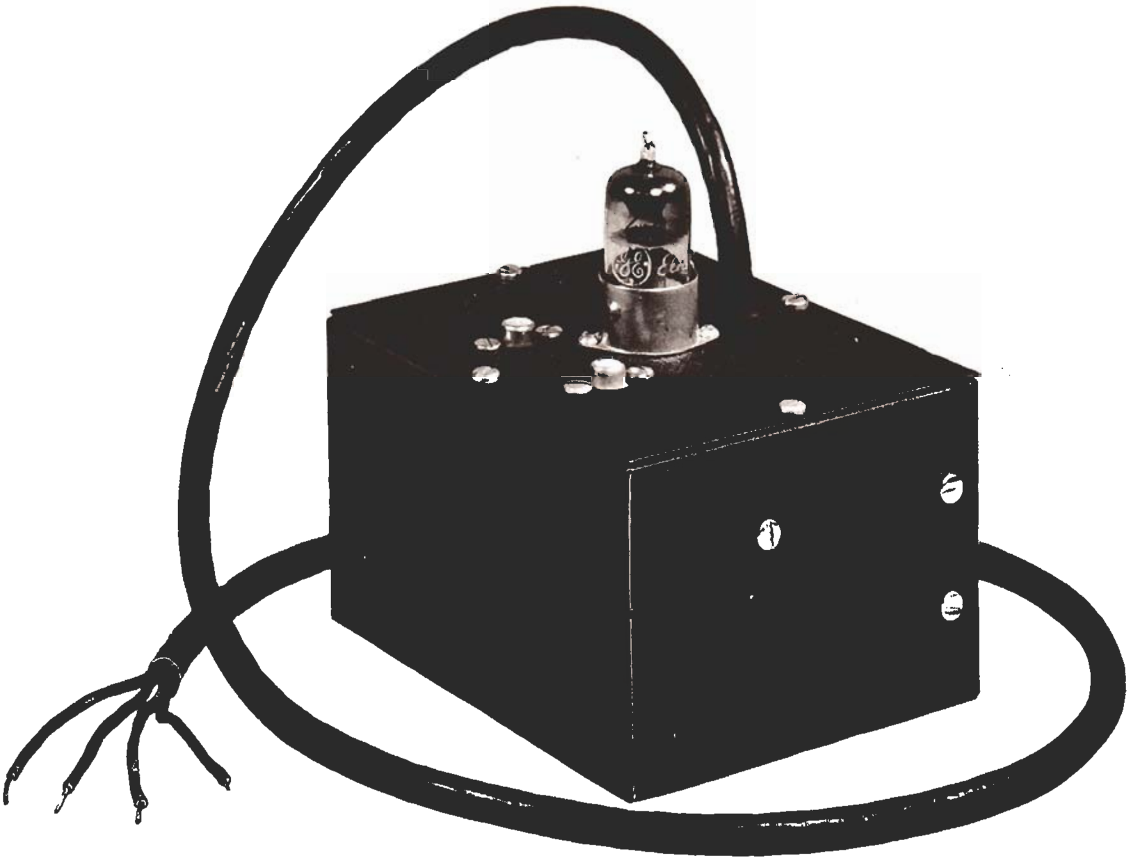
4. SUBSTITUTING OTHER RECTIFIERS FOR "W" -- Tests have indicated that germanium, selenium and silicon rectifiers exhibit the same logarithmic characteristic as copper oxide rectifiers, but to a lesser extent. Thus, they will not perform as efficiently in this circuit over such a wide compression range (about 40 db with the copper oxide rectifiers).

5. INCORPORATING COMPRESSOR INTO EXISTING AUDIO GEAR -- The basic compressor circuit -- C_4, C_5, R_7, R_8, R_9, T_1 and W -- may be connected into an existing speech amplifier circuit. However, the frequency response of the stages preceding the compressor should be tailored to attenuate frequencies below 300 cycles. Also, extra components (low-pass RC filter formed by R_8, R_9, C_5, C_6, C_7, C_8, etc.) should be added to the following stages to attenuate any harmonics generated in the compressor circuit.

6. LOGARITHMIC COMPRESSOR ON SSB TRANSMITTERS -- Several radio amateurs with filter-type SSB transmitters have reported that this type of compressor can be used successfully. Audio harmonics generated in the usual type of clipper can cause severe ringing in the sideband filter circuits, with disastrous effects on signal quality. Check the amount of compression with an oscilloscope so that the SSB exciter and linear amplifiers are not overloaded, since adding the compressor will increase the average power input of a linear SSB transmitter from TWO to TEN times.

HIGH ATTENUATION LOW-PASS AUDIO FILTER

From March-April, 1955



This audio filter for receiver or speech amplifier uses inexpensive unshielded coils plus a few of W2KUJ's slick tricks to obtain an attenuation slope approaching that possible with high-priced toroid coils.

—*Lighthouse Larry*

LOW-PASS AUDIO FILTER

A sharp cut-off low-pass filter is a great help in eliminating the annoyance of heterodynes and noise beyond the range necessary for completely satisfactory 'phone reception. The filter described here is an inexpensive and highly effective weapon in the fight against QRM. Used in the speech system of a transmitter, this filter reduces the spectrum space occupied by the signal, while actually increasing the effectiveness of the transmission. It is connected as shown in Fig. 1.

Because the filter is intended for a variety of applications, a vacuum tube is employed to provide high input and low output impedance. Thus, all the requirements of impedance matching for the passive elements of the filter are satisfied internally and are not disturbed when the filter is interposed between a wide variety of devices.

The design cut-off frequency of the filter pictured on the cover of this issue of G-E HAM NEWS is 3000 CPS, a figure generally considered adequate for voice communication. Design data is given for the prototype low-pass filter in "Thumbnail Theory" for those who want to design their filter for a different cut-off frequency. It is suggested that the 3000 CPS cut-off design be used unless it is certain that a different cut-off point is required for some specific application.

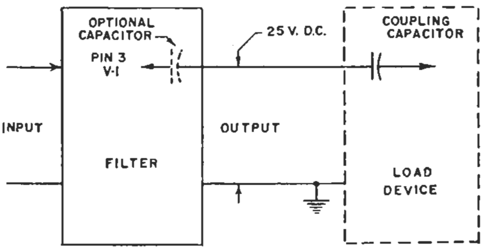


Fig. 1 Diagram for connecting the low-pass filter to the input and load devices.

CONSTRUCTION

The entire filter is housed in a 3 x 4 x 5-inch utility box drilled as shown in Fig. 2. The tube socket, input and output jacks are mounted on one cover. Although the filter elements are not in "cramped" space, a certain amount of clearance is required between coils in different filter sections. The circuit diagram and parts list are shown on page 5.

The six 125-mh coils are mounted on the aluminum brackets shown in Fig. 3 with 6–32 brass machine screws 1 inch long which pass through the centers of the coils. The brackets are then fastened to the 3 x 4-inch ends of the box, as pictured in Fig. 4. Note how L_1 and L_6 are mounted with respect to the other coils. All the wiring between the coils, condensers C_1, C_2, C_3 and the 2400-ohm terminating resistor is done with the covers removed. Attach 4-inch leads to the input end of L_1 and the output end of L_6 for later connection to the tube socket.

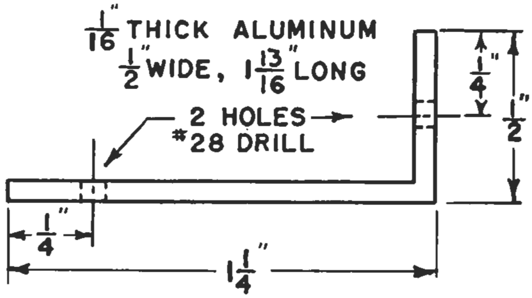


Fig. 3 Mounting brackets for the coils.

The brackets holding L_1 and L_6 should not be securely tightened until after performance tests are completed. Do not rely on the steel box to provide the ground path indicated in the circuit diagram. Instead, run a lead to the grounded ends of C_1, C_2 and R_4, and bring this lead out to the ground points on the top cover. Heater and plate power are supplied through a four-conductor cable anchored to the cover.

TESTS

After a wiring check, heater and plate voltage can be applied. Approximately 25 volts DC should appear across the output resistor. If the correct values of inductors and capacitors have been used, the performance

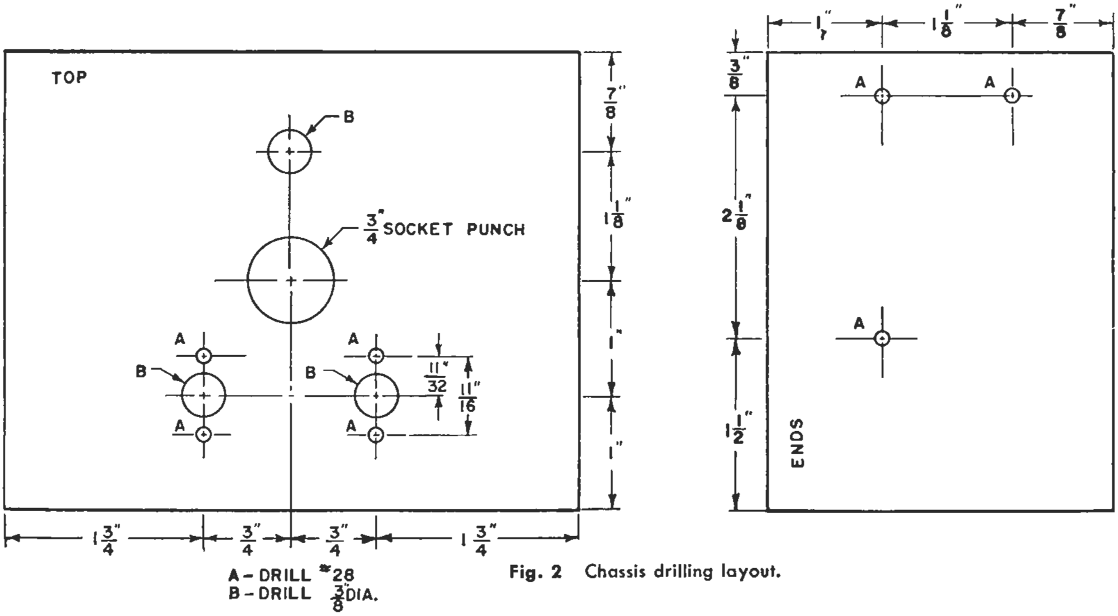


Fig. 2 Chassis drilling layout.

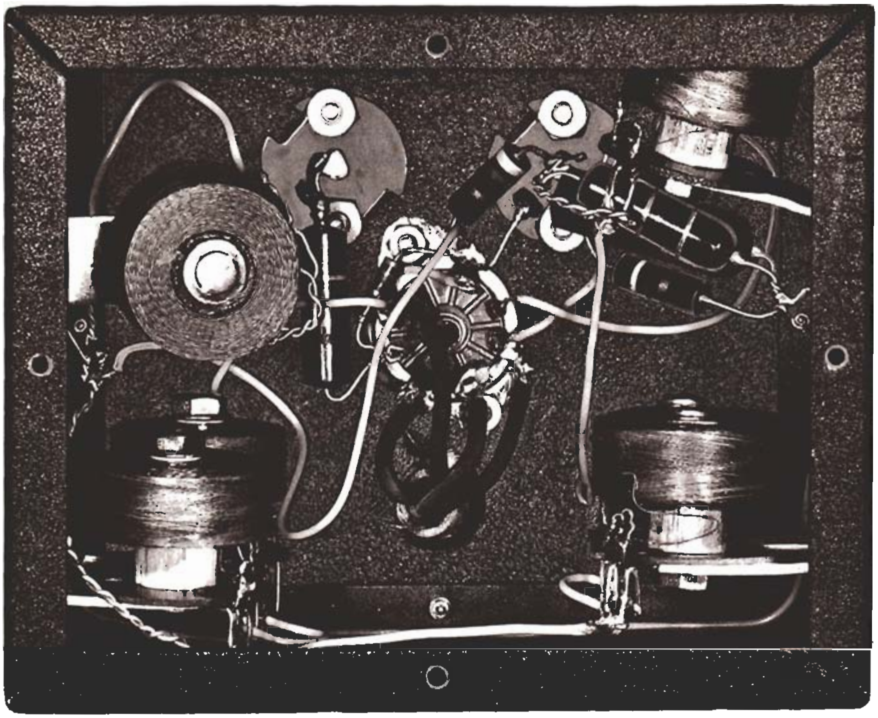


Fig. 4 Bottom view of filter showing socket placed with pins 1 and 9 toward phono jacks. Solder lugs under coil bracket mounting screws are used as ground tie-points for C_1, C_2 and C_3.

will be that shown by the curve "A" of Fig. 6, at least to an attenuation of 30 db without any further work on your part. Usually an improvement in attenuation at frequencies higher than 4000 CPS can be made by orientation of L_1 and L_6, if suitable measuring equipment is available. Tests made by ear alone are not sufficiently reliable to warrant the effort. A reliably calibrated audio oscillator covering a range from 100 to 10,000 CPS at an output voltage of about 10 volts RMS, and an output indicator covering a range of at least 60 db (1000 to 1 in voltage) are required.

In case orientation of L_1 and L_6 through a few degrees does not allow an attenuation of 60 db or more to be obtained at 6000 CPS, reversal of connections to either L_1 or L_6 (but not both) should allow the performance shown in curve "B" to be equalled or surpassed in the region of high attenuation. The final adjustment of the filter model shown in the illustrations was obtained by setting the test frequency at 7000 CPS and bending the brackets holding L_1 and L_6 for minimum output. Tests with an oscilloscope revealed that the minimum was really a null at 7000 CPS and that the measured output 85 db below the reference level was hum and noise. Beyond 7000 CPS the output rose to about 70 db below reference level and dropped slowly above 10,000 CPS. The insertion loss of this filter is 7 db; that is, the output voltage at 100 CPS is 7 db less than the input voltage. This loss is a consequence mainly of the resistance of the choke coils used.

APPLICATION

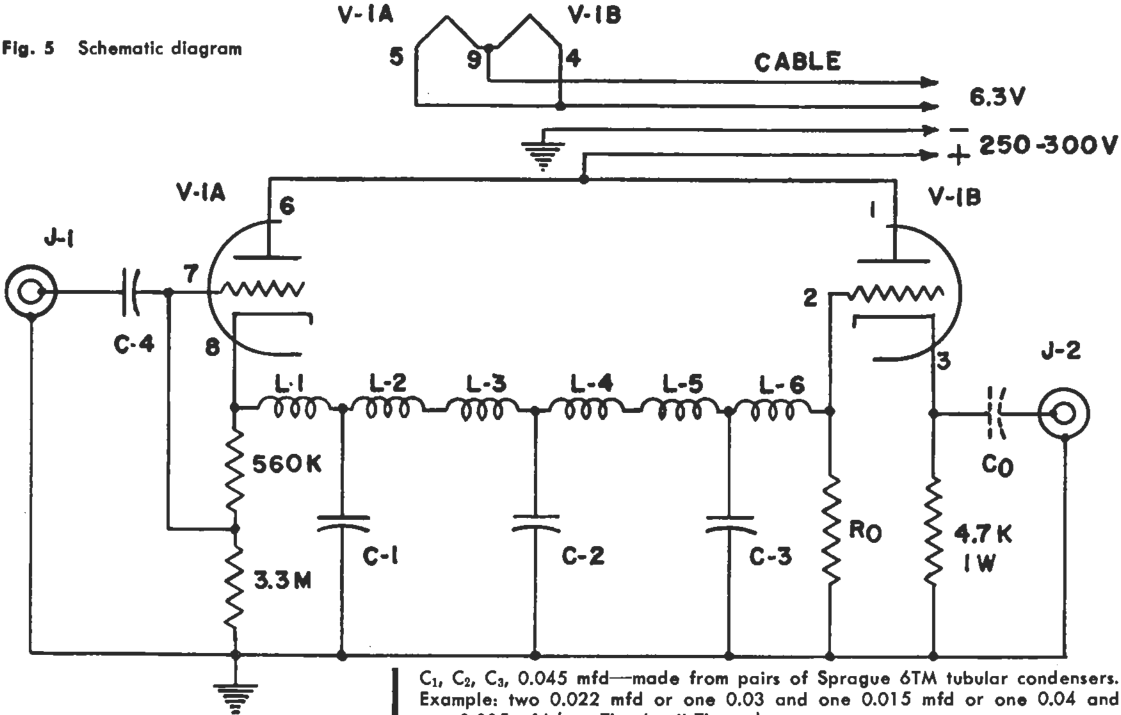
The maximum operating level for the filter is 10 volts RMS at the input. Operation at higher levels will introduce distortion due to overloading of the input triode. Practical operating levels will range between 1 and 10 volts. Operation at lower levels will, of course, degrade the signal-to-hum ratio. It will be observed that the hum level in the output is determined by the amount of stray magnetic field in the vicinity of the filter since the coils are not magnetically shielded. A power supply is not included as part of the filter for this reason. Ordinarily, the small amount of heater and plate power required can be borrowed from other apparatus with which the unit is used. If excessive hum is experienced, try moving the filter to a more favorable position, or orienting it for minimum hum pickup.

As a receiving accessory, the filter is inserted in an audio circuit where the operating levels are within range. In most receivers, the output of the first audio stage will provide a suitable signal level for the filter input. When used with the Signal Slicer (G-E HAM NEWS, Volume 6, No. 4) the filter should be inserted between the slicer output and its succeeding audio amplifier. Note that the 4700-ohm output resistor should not be short-circuited by the device into which the filter operates. A coupling capacitor with reactance equal to one-tenth the input impedance of the load at the lowest desired frequency should be provided at the input to any such load (see Fig. 1). Such a capacitor can be incorporated as part of the filter unit to avoid mistakes.

Whether the crystal filter in the receiver is used or not, this filter will improve CW reception somewhat even though the bandwidth is greater than needed for that application. The improvement obtained will depend on the characteristics of the receiver and the particular QRM problem encountered.

For use as a bandwidth control in transmission, the filter is inserted in the audio circuits at a point where the operating levels are suitable. The above precautions regarding the load circuit should be observed. When

Fig. 5 Schematic diagram



PARTS LIST

C_1, C_2, C_3, 0.045 mfd—made from pairs of Sprague 6TM tubular condensers. Example: two 0.022 mfd or one 0.03 and one 0.015 mfd or one 0.04 and one 0.005 mfd (see Thumbnail Theory).
C_4—0.01 mfd, 600 volts.
C_0—Capacity dependent on load impedance, see text (Application).
J_1, J_2,—Cinch shielded phono jacks.
L_1,—L_6—125 mh RF chokes (Meissner 19-6848).
R_0—2400 ohms, 1 watt ±5%.
V_1—12AT7 tube.
All resistance values in ohms, ½ watt ±20% except as noted. K=1000.

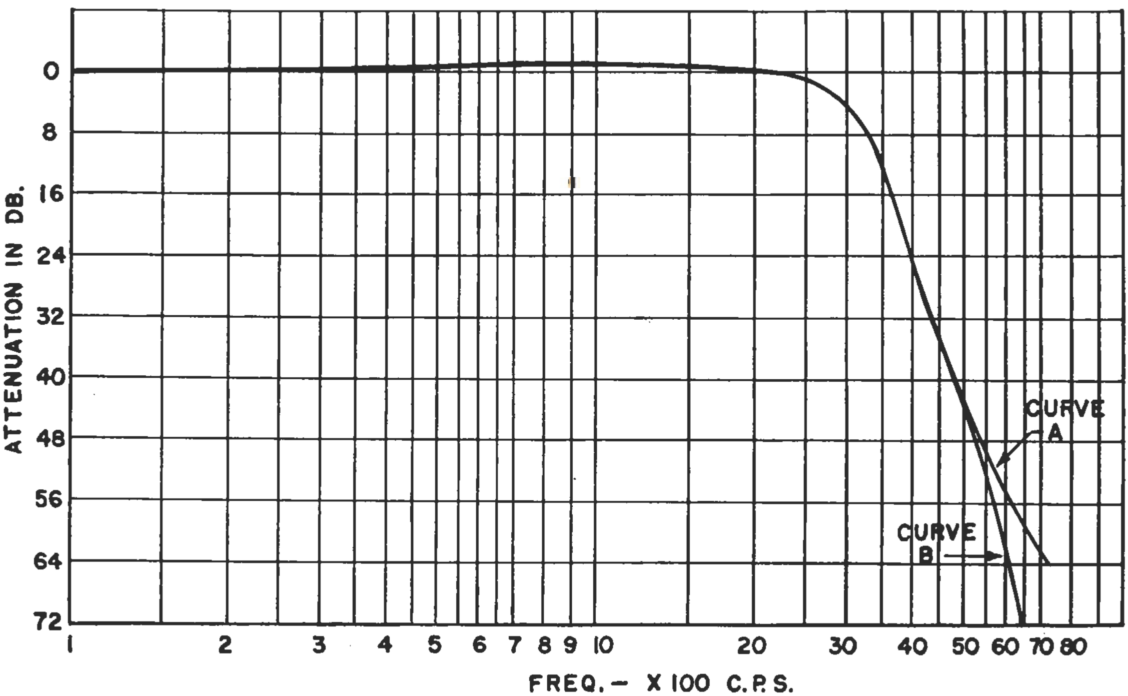


Fig. 6 Response curves with and without coils oriented.

used in conjunction with the SSB Jr. exciter (G-E HAM NEWS, Volume 5, No. 6) the filter output can connect directly to the audio input jack of the exciter if an 0.01 mfd coupling condenser is inserted in either the filter output or exciter input.

The filter characteristics do not provide for attenuation of low frequencies. Where it is desired to tailor the audio response of the transmitter, this may be done in the circuits either preceding or following the filter. When low-frequency attenuation is introduced after the filter, hum pickup in the filter itself will be attenuated. (See G-E HAM NEWS, Volume 4, No. 4, for simple means of introducing low-frequency attenuation in speech amplifier circuits.)

The "dyed-in-the-wool" experimenter will find many other applications for a handy sharp cut-off filter such as the one described here. Even though the filter is normally used in only one place (say as part of the receiver setup) it will be found convenient to provide input and output jacks so that the device may be patched into other apparatus as the need occurs. In this way a single filter can be made to serve a variety of uses.

THUMBNAIL THEORY

The design of filters can not be covered very thoroughly in a few paragraphs. For those who want some background information on filters the following will be of interest.

The basic filter section used in the device described in this issue is called the "constant K prototype," shown in Fig. 7. Any number of these sections may be joined together for greater attenuation beyond the cut-off frequency. When this is done, the internal sections can be considered as either π or T sections. A multiple section filter is called a "composite" filter. In the ideal case, a constant K filter must be driven by a source having an internal impedance equal to the characteristic impedance of the filter section and

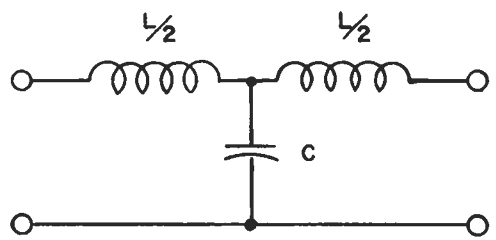
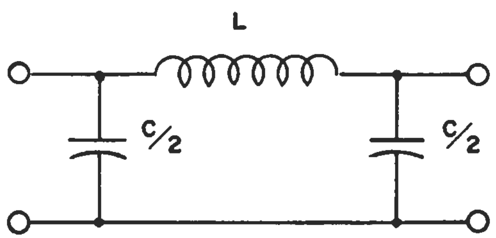


Fig. 7 Above is a T structure filter section. When several T's or π's are joined in a complete filter, the internal sections lose their identity.

Fig. 8 Below is a π structure filter section.



the filter must be terminated by the same impedance. Ideally, too, the filter elements should be perfect reactances. Practically speaking, the characteristic impedance varies throughout the pass-band of the filter. Of course, the filter elements do have loss (they are not perfect reactances) so that other considerations enter into the design of filters. Even when perfect filter elements are assumed, the variation of characteristic impedance within the pass-band presents a problem that is solved partially by more complex circuit arrangements known as "M-derived" filters.

In filter design, as in most things, a compromise must be made between performance and complexity, or cost. In our case, certain liberties were taken with classical filter theory to provide acceptably good performance with basically straightforward and simple circuits. A low source impedance is provided by the cathode follower input arrangement shown in the schematic diagram, while the terminating impedance is a resistor of a constant value. These departures cause minor variations of the attenuation within the pass-band. Fortunately, these variations are partially smoothed out by the loss in the filter coils and the approach to ideal operation is thereby improved. Non-ideal filter elements can be used with considerable saving in cost and a less complex filter arrangement can be built. The loss in the coils accounts for the bulk of the "insertion loss" mentioned earlier. About 2 db of the insertion loss is accounted for by the two tube sections used.

The composite filter in this article comprises three identical T sections joined together. Since a low driving impedance is used, no advantage could be achieved by a π structure. Rather than select a certain characteristic impedance and then prune commercial coils to necessary values in order to provide the desired cut-off frequency, the design equation for inductance *per section* was solved for R_0, the low-frequency characteristic impedance. Thus:

(1) $R_0 = \pi f c L$, where fc is the cut-off frequency and L is twice the inductance value obtainable. When fc = 3000 CPS, then L = 0.25 henry.

(2) $R_0 = \pi \times 3000 \times 0.25 = 2360$ ohms. Keeping fc at 3000 CPS, and using $R_0 = 2360$ ohms

(3) $C = \frac{1}{\pi f c R_0} = \frac{1}{\pi \times 3000 \times 2360} = 0.000000045$ farads.

Thus C = 0.045 microfarads.

Equations (1) and (3) can be used in designing low-pass filter sections for other cut-off frequencies if desired. A filter is said to "cut-off" when its attenuation reaches 3 db.

The coils used in the sample filter had an inductance value of 0.125 henry each. The measured Q at 1000 CPS was 2.20. The total value of 0.25 henry required per section is twice the value of the individual coils obtainable. The filter capacitors were made up of two selected commercial plastic-encased paper capacitors connected in parallel to provide the calculated value of 0.045 ufd. The individual capacitors were checked for value and paired for as nearly matched composite values as possible, as well as adherence to the design value required.

The sharpness of cut-off obtained with this filter is greater than that indicated by classical filter theory when constant resistive source and load impedances of the value R_0 are used. This greater attenuation is paid for by the irregularities shown between 1000 and 2500 CPS, a really small price indeed. The additional attenuation obtained beyond the 40 db point by coupling between L_1 and L_6 to provide "infinite" attenuation at 7000 CPS serves to increase the slope of the characteristic between 4000 and 7000 CPS at the expense of smaller attenuation beyond about 10,000 CPS. Although this actual difference is measurable, its practical significance for most applications is very small.

THE FIRST GADGET RACK ACCESSORY is a combination keying monitor, modulation indicator and field strength measuring instrument.

CONTINUOUSLY CHECKING your transmitter signal—and your *fist* too—is easy with this versatile unit. It requires only three tubes and two germanium diodes. A plate and post chassis, shown in the side view, Fig. 1, automatically provides a thru-panel mounting for the 6E5 indicator *eye* tube.

The signal to be monitored is fed into the unit from an external pickup antenna on pin 10 of the interconnecting cable system, as shown in the schematic diagram, Fig. 2. A 100-ohm potentiometer adjusts the signal level applied to the 6BE6 mixer tube. The position of the function switch, S_1, determines the operation of the remaining circuits, as follows:

CW—An NE-51 neon lamp relaxation oscillator generates an audio tone which is mixed with the RF signal from the transmitter. This produces a modulated RF signal in the 6BE6 plate circuit, tuned to the transmitter frequency.

After detection by a 1N34 diode, the resulting audio signal is amplified in the left-hand 12AX7 triode and appears in the headphone circuit (pin 5 of the bus-bar system). Signals from the station receiver, applied on pin 11, are also fed into the headphone circuit by the right-hand 12AX7 triode. However, whenever the transmitter key is pressed, rectified RF voltage from the 1N34 is applied to this stage as a negative bias, disabling it.

Thus, receiver audio is present in the headphone circuit when the key is *up*, and the NE-51 audio tone is heard when the key is *down*. This function is similar to the popular *Monitone* circuit[1].

CARRIER LEVEL—In this position of S_1, an RF signal from the transmitter results in application of negative bias from the 1N34 to the grid of the 6E5 *eye* tube. This causes the unlighted portion of the circular fluorescent target on the end of the 6E5 to narrow or close entirely, indicating relative carrier level.

MODULATION—In this position of S_1, modulation on a transmitter signal, detected by the 1N34 diode, appears in the headphone circuit. This audio signal also is rectified by a second 1N34 (located between S_1 and the 6E5 in Fig. 2), applied as a negative bias on the 6E5 grid and causes the *eye* to close in accordance with the modulation on the transmitter signal.

THE MODEL SHOWN was constructed on a 2½-inch-wide panel and a 2¼-inch-wide chassis plate. Parts locations on the chassis layout diagram, Fig. 3, are not critical and may be changed to suit available components. Good construction practice—short leads, isolation of signal and AC power circuits, related components grouped together, etc.—should be followed, however.

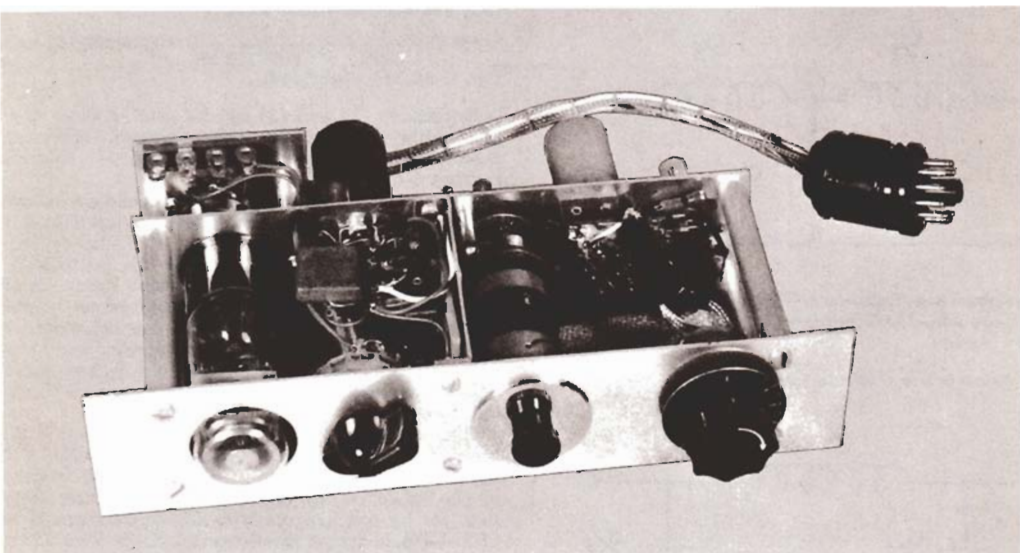


FIG. 1. SIDE VIEW of the COMBO MONITOR unit. Corner posts connecting the panel and chassis are 3 inches long. Those for the 2 x 2¼-inch mounting plate for the 1-megohm potentiometer are 1½ inches long. All posts are tapped for 6–32 screws at both ends.

PARTS LIST

C_1....Midget mica-insulated trimmer capacitors connected across L_1 in each coil; see COIL TABLE for values.

C_2....100-mmf mica (Or, 75 to 150 mmf, see text, page 8).

NE-51....1/25th-watt neon glow lamp; requires miniature bayonet socket.

P_1....Male 11-pin octal plug (Amphenol 86-PM11).

S_1....2-circuit, 3-position, single section, non-shorting tap switch (Mallory 3223J).

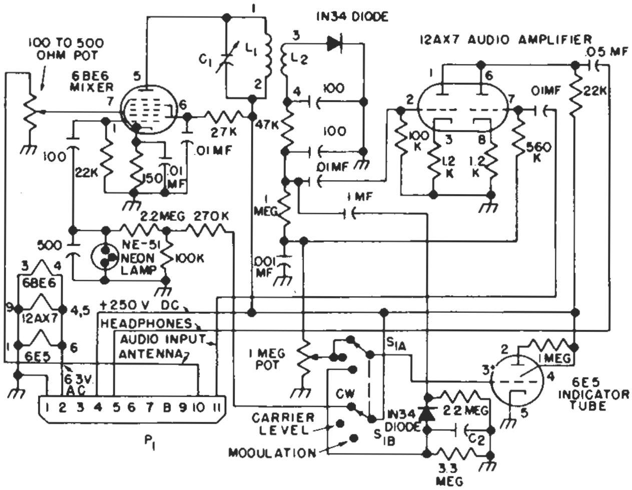


FIG. 2. COMPLETE SCHEMATIC DIAGRAM for the COMBO MONITOR. Chassis grounds in the model were made at the points indicated. All capacitances are in mmf; all resistors ½-watt composition, unless otherwise specified.

ADJUSTMENT IS SIMPLE, once all circuits in the COMBO MONITOR are working properly. Plug in a coil for the band on which the transmitter is operating before applying power to the unit. Modulate the transmitter 100 percent (check this with an oscilloscope, borrowed or otherwise), turn S_1 to *MODULATION* and adjust the 100-ohm signal lever potentiometer until the 6E5 *eye* barely closes.

Remove the modulation from the transmitter, turn S_1 to *CARRIER LEVEL,* and adjust the 1-megohm potentiometer so that the 6E5 *eye* just closes, but does not overlap. The monitor is now calibrated to indicate 100-percent amplitude modulation of a transmitter. The 1-megohm potentiometer can now be locked in position.

Each time the monitor is used on a different band, simply turn S_1 to *CARRIER LEVEL* and adjust the 100-ohm signal level control so that the 6E5 *eye* barely closes. Then return S_1 to the *MODULATION* position and the monitor is ready for use on a modulated signal.

[1] J. W. Paddon, "The Monitone," September, 1948, *QST*, page 22; also C. V. Chambers, "The Monitone—Model 1951B," May, 1951, *QST*, page 29.

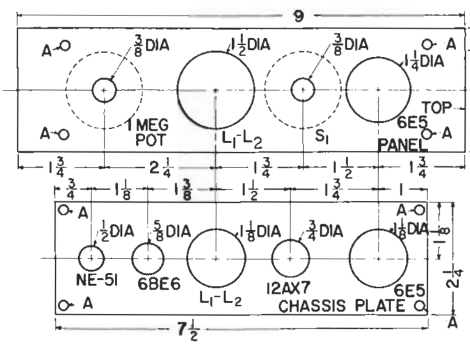


FIG. 3. PANEL AND CHASSIS PARTS LAYOUT used for this model. Small holes for socket and terminal strip hardware are not shown and should be located from those parts.

COIL TABLE—COMBO MONITOR

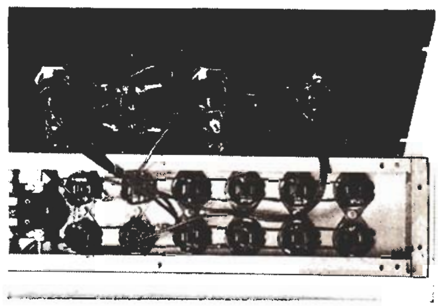
All coils wound with No. 24 enameled wire on 1-inch diameter, 4-prong coil forms (Millen, No. 45004; ICA, No. 1108B). On 3.5- and 7-megacycle coils, L_2 is wound over the pin-2 end of L_1, with a layer of plastic insulating tape between. On 14-, 21- and 28-megacycle coils, L_2 is wound next to pin-2 end of L_1.

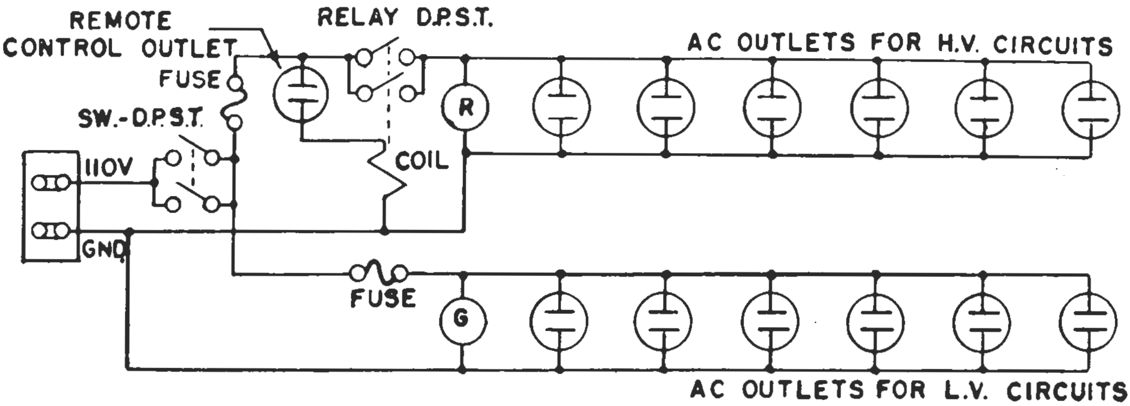
BAND (MC)	L_1 uh	L_1 turns	L_1 length	L_2 turns	C_1 (mmf)
3.5	42	52 (closewound)	1⅛"	16	4—50
7	13	24 (closewound)	½"	12	4—50
14	7	16 (closewound)	⅜"	8	3—12
21	3.6	12 (spaced wire dia.)	⅝"	6	3—12
28	2	7 (spaced wire dia.)	⅜"	4	3—12

Somehow the problem of how to switch the rig and associated equipment on and off seems to sneak up on a fellow unsuspectingly. He concentrates on his transmitter, receiver, converters, VFO and the other pieces of equipment and when he gets them all working suddenly realizes he doesn't have any way to operate them without flipping a dozen or so separate switches.

Then he has to scramble around hunting in handbooks and magazines and calling up his friends to get ideas for a control unit of some sort. Of course, it's not a difficult problem, and there are endless ways of solving it.

We present this solution—found in the shack of W2GYV—as one more suggestion to add to the pile. This is a 7-inch control panel with a 4 x 17 x 3-inch chassis mounted as shown.





The circuit is simple and provides for remote control via the a-c type female outlet on the front of the panel. On the rear of the unit are two rows of a-c outlets—both supplying 110-volt a-c. The bottom row of six outlets is controlled by the front panel switch and is used for filament circuits in other pieces of equipment. The top row of outlets is controlled by the same switch plus the relay, and offers 110-volt a-c for the high-voltage plate transformers of various pieces of equipment.

An interesting feature of the circuit is that the relay coil is connected in the grounded side of the 110-volt a-c circuit. This method of connecting the coil eliminates any possibility of the relay being actuated if the hot lead in the remote control cable should accidentally become grounded. Incidentally, the relay used here is a double-pole type to provide a wide margin of current-carrying capacity and to halve the possibility of poor contact because of dirt or corrosion. A single-pole relay can be used.

The toggle switch shown is a heavy-duty, double-pole type to insure plenty of current-carrying capacity. The fuses used should be chosen to just carry the total current that will be drawn in their respective circuits.

The photographs show the construction clearly. Note that the mounting plates for the a-c outlets are overlapped to fit neatly in the chassis. Nothing in the construction is critical and the builder can make whatever variations are necessary to suit his purpose.

One excellent feature to add would be an interlock switch in series with the remote control outlet.