A Filter Design for the Single-Sideband Transmitter

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SINGLE-sideband transmission has been in use on the amateur bands for well over a year, and its theoretical advantages over a.m. and n.f.m., both in reducing QRM and its ability to "get through," have been proved in practice. Although there is room for improvement in receiver stability and selectivity, this has not proved such a handicap as it first seemed.

The greatest obstacle to greater single-sideband activity is the need for a simple and inexpensive means of converting existing a.m. transmitters to single-sideband operation. While it has been used commercially for many years, single sideband is new to most amateurs. Like any new technique, it appears complicated at first glance. However, many excellent articles [1] have appeared in *QST* and other publications, for anyone who is still unfamiliar with the basic principles.

Briefly reviewing: One basic method of producing single sideband, termed the "phasing" method, eliminates carrier and undesired sideband by employing two balanced modulators with 90-degree carrier and audio networks.[2] This system enables the sideband to be produced directly at the desired output frequency. While this has certain advantages, there is little assurance that the necessary high degree of phase and amplitude balance will be maintained over long periods of time. There is also difficulty in determining with simple test equipment whether the undesired sideband has been sufficiently suppressed.

In the filter method of generating single sideband, a double sideband is first generated in a balanced modulator (where the carrier is eliminated), and the filter removes the undesired sideband by "brute force." While this method does not have the "finesse" claimed for the "phasing" method, it is positive, and requires no critical adjustments. Since the filter is a passive network, sideband suppression is not affected by other circuit variations, tube gains, etc.

Filters using only inductors and capacitors are practical only at frequencies in the order of 10 to 50 kc., and the sideband must be obtained at some low frequency and heterodyned to the desired output frequency. This is not a serious handicap, and enables the output frequency to be varied without disturbing the sideband-generating portion of the circuit. Contrary to the statement made by some, the selection of either upper or lower sideband is simple, requiring only a frequency change of one of the oscillators.

The block diagram of a practical single-sideband transmitter is shown in Fig. 1. The selection of upper or lower sideband is accomplished by switching the frequency of the second oscillator. In the notation of Fig. 1, "USB" and "LSB" indicate the position of sideband at the points noted with respect to the input speech frequencies. This is not to be confused with the particular sideband of the second oscillator that is selected by the second i.f. Careful study of Fig. 1 will make this clear.

Although not directly indicated in Fig. 1, the

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[1] Goodman, "What Is Single-Sideband Telephony?" *QST*, Jan., 1948; Norgaard, "What About Single Sideband?" *QST*, May, 1948.

[2] Norgaard, "A New Approach to Single Sideband," *QST*, June, 1948.

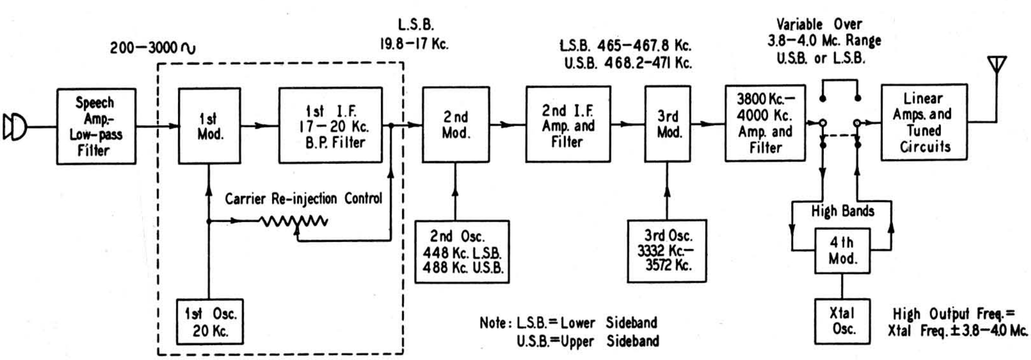
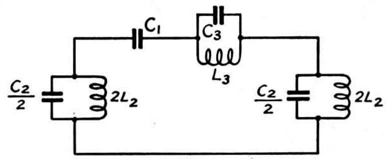


Fig. 1 **— Block diagram of a typical single-sideband suppressed-carrier transmitter or exciter. The equipment inside the dashed area is described in this article.**

requirements of the various filters might be briefly reviewed. The first i.f. filter must select a band of frequencies from about 19.8 to 17 kc. and have high attenuation to frequencies of 20.2 to 23 kc. (the other sideband). The second i.f. selectivity must be such that it will greatly attenuate the frequencies of the unwanted sideband generated in the second modulation process. This unwanted sideband will be removed by twice the frequency of the first i.f. (34 to 40 kc. for 17–20 kc. first i.f.). The second-oscillator frequency must also be prevented from being transmitted. A balanced modulator for the second modulator will remove most of this undesired signal but is not to be relied upon for complete elimination; therefore, the second i.f. should also have considerable attenuation at ±20 kc. Coupled tuned circuits of conventional i.f. transformer design are satisfactory at the frequencies chosen.

The third i.f. requirements are similar to those of the second i.f. except that, with the frequencies chosen, the selectivity requirements are more lenient. It would be quite practical to employ a mixer (modulator) not of the balanced type, for the third modulator. Further notes on the complete exciter or transmitter are beyond the scope of the present article, as the first i.f. filter is considered to be more difficult to the average constructor and is deserving of full consideration.[3, 4]



$$C_{1K}=\frac{f_2-f_1}{4\pi f_1 f_2 R}\;;\; L_{2K}=\frac{(f_2-f_1)R}{4\pi f_1 f_2} \qquad h=\sqrt{\left(1-\frac{f_1^2}{f_{2\infty}^2}\right)\left(1-\frac{f_2^2}{f_{2\infty}^2}\right)}$$

$$C_1=a\,C_{2k}\;;\; C_2=m_1\,C_{2k}$$

$$C_3=\frac{1-m_1^2}{4h}\cdot C_{2k} \qquad a=\frac{(1-m_1^2)f_{2\infty}^2}{4f_1 f_2}$$

$$L_2=L_{2k}$$

$$L_3=\frac{h}{a}\cdot L_{2k} \qquad m_1=\frac{f_1 f_2}{f_{2\infty}^2}+h$$

Fig. 2 — **The basic *m*-derived pi section used in the filter.**

The primary purpose of this article is to describe in detail the construction of a highly-selective first i.f. filter that can be built at a reasonable cost and with a minimum of special test equipment.

[3] The general technique is shown by Nichols, "A Single-Sideband Transmitter for Amateur Operation," *QST*, Jan., 1948.

[4] In the original manuscript, Mr. Berry showed the 2nd oscillator (Fig. 1) on either 448 or 485 kc. It was changed as shown because this maintains the output (suppressed) carrier on the same frequency when transmitting either upper or lower sideband, and it is a little easier.

Filter Design

In considering the design of this filter, quality of components and desirable characteristics were of first consideration; low cost and ease of construction were achieved by selection of the type of filter sections and impedance transformations. Sharp cut-off is restricted to the high-frequency edge of the passband, concentrating the attenuation where most needed, and resulting in a minimum number of inductors. This filter is designed for selection of the lower sideband, but since the position of the sideband may be altered at the output of the transmitter in a succeeding modulator stage, this is no handicap.

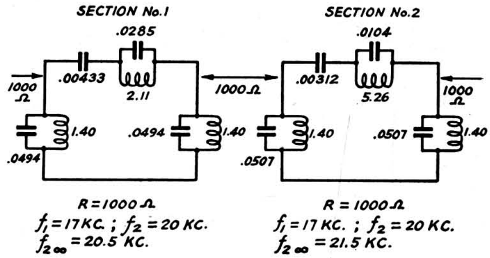


Fig. 3 — **Component values of the individual pi sections of the filter. Values are in μfd. and mh.**

A figure of approximately 40 db. reduction of the undesired sideband was selected as a practical value. It is believed that values much lower may tend to limit operation on the adjacent channel (when sufficiently selective receivers are in use). Values much over 40 db. would probably not be worth while even if a greater ratio were obtained at the output of the filter, because intermodulation in succeeding stages of the transmitter is likely to introduce spurious beat products of low intensity. (Note: In any single-sideband transmitter, improper amplifier bias and overloading in the linear amplifiers is to be avoided as the effect is similar to an overmodulated a.m. transmitter, with its resultant splatter.[5])

A bandwidth of 2800 to 3000 cycles has proved satisfactory for commercial communication and is thought to be a practical one for amateur use.

A frequency band of 17 to 20 kc. was chosen in preference to one of lower frequency to reduce the selectivity requirements of the second i.f. filter, as previously noted. This rather high frequency (for a single-sideband filter) also has the feature of lower component values, lowering cost and making hand winding of the inductors feasible.

The filter consists of two *m*-derived pi sections of the type shown in Fig. 2. This type of section has one frequency of infinite rejection on the high-frequency side of the passband. By using two sections, one with the rejection frequency at

[5] Reque, "Linear R.F. Amplifiers," *QST*, May, 1949.

20.5 kc. and the other at 21.5 kc., the resultant attenuation on the high-frequency side is quite high. When the two sections are combined, the inductors and capacitors at the junction may be combined, to reduce to five the total number of inductors in the complete filter.

The input and output impedance characteristics are the same as that of the mid-shunt constant-k type of filter of the same cut-off frequency. This sort of termination impedance is economical and works well either directly from a ring modulator or resistance terminations.

The design impedance R of 1000 ohms was selected to give desirable component values and desirable input and output impedances obtained by transformer action in the end inductors.

The resultant values calculated from the design formulas of Fig. 2 for each section are given in Fig. 3, and in Fig. 4 for the two sections combined.

To those who may wish to calculate similar filters, note that if sections are to be joined, the design impedance and the cut-off frequencies must be the same for both sections, although the frequencies of infinite attenuation may be different.

The filter of Fig. 4 might now be constructed, and if proper components are available, the insertion loss between 1000-ohm resistive impedances would be approximately that of Fig. 5. A low dissipation factor (high Q) is necessary in most of the components to obtain the required characteristics. Resistive losses internal to the filter not only will cause a greater loss at all frequencies but will "round off" the edges and prevent the rapid rise of attenuation needed just outside the passband.

Except in the case of C_1 and C_7 (Fig. 4), mica or other low-loss types of capacitors are necessary for proper filter action. C_1, C_4 and C_7 are large values for mica capacitors and would be expensive. Ordinarily they would have to be made from a large number of paralleled units. C_1 and C_7 appear directly in parallel with the terminating resistances and it has been found that good-quality paper capacitors are satisfactory here.

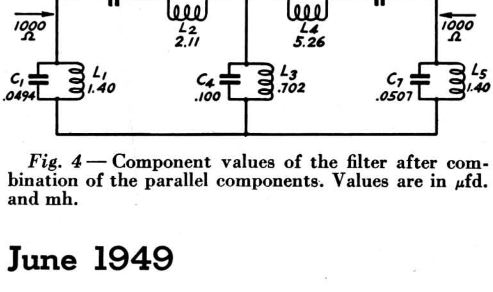


Fig. 4 — **Component values of the filter after combination of the parallel components. Values are in μfd. and mh.**

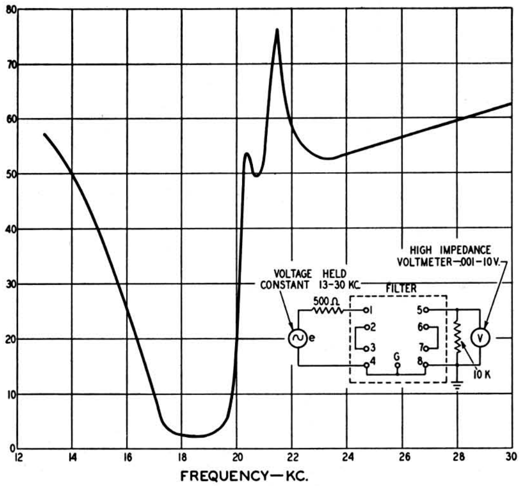


Fig. 5 — **Attenuation characteristic of the filter shown in Fig. 4. The test set-up diagram refers to Fig. 6.**

C_4 is located internally and must be of low-loss type for best results. It is possible, however, to use impedance transformation at L_3C_4 and permit a smaller mica capacitor to be used for C_4 at the expense of a larger value for L_3. The method of impedance transformation employed also permits a relaxation of the capacitor tolerance. Any reasonable value may be used for C_4 provided the inductor is adjusted to the correct antiresonant frequency. The correct impedance may then be regained by tapping L_3. With inductors of high coupling between turns, the proper point of tapping is such that the inductance between common and tap is approximately that of the value of L_3 before transformation. The modification with the more desirable values is shown in Fig. 6.

Component Tolerances

In the filter of Fig. 4, the tolerance of some of the elements is quite critical, particularly that of the series arms. It has been found in the design of filters of this type that the tolerance of LC ratios is not particularly critical provided the correct resonant and antiresonant frequencies are maintained. Practically, this leads to the selection of capacitors to a tolerance of ± 5 per cent or better, and resonating each LC circuit to the correct frequency by turn adjustment of the inductor. The maximum possible error of 10 per cent in the impedance match between junctions of the filter arms is not serious. Greater tolerances will cause a "ripple" in the passband and other deviations from the desired characteristics. In following this procedure, note that the series arms of the filter have both a resonant and an antiresonant frequency, with the inductor as a common element for both. Obviously, the inductor could not be adjusted independently for both frequencies. To permit this desired independent adjustment, a tapped-inductor arrangement is used.

Considering the series arm C_2-L_2-C_3, C_3 is selected with a tolerance such that it will always be larger than the calculated nominal value. L_2 may then be adjusted with this new value of C_3 to the correct antiresonant frequency and will have fewer turns than the original calculated value of L_2. Leaving C_3 connected across the exact number of turns necessary for antiresonance, turns may be added to L_2 until the entire combination of L_2, C_3 in series with C_2 and the added winding of L_2 will series-resonate at the correct frequency. The exact value of C_2 will set the impedance of the entire arm, and $\pm$ 5 per cent is permissible.

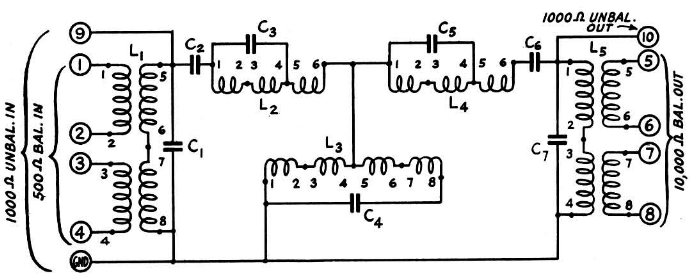


Fig. 6 **— Revised filter of Fig. 4, with provision for balanced or unbalanced input and output.**

C_1 — 0.049 μfd. ±5%, paper.
C_2 — 0.0043 μfd. ±5%, mica.
C_3 — 0.03 μfd. + tol., mica.
C_4 — 0.03 μfd. ±20%, mica.
C_5 — 0.011 μfd. + tol., mica.
C_6 — 0.0031 μfd. ±5%, mica.
C_7 — 0.051 μfd. ±5%, paper.
L_1 (1–4) — 0.7 mh., 33 turns No. 26, bifilar.
L_1 (5–8) — 1.4 mh., approx. 47 turns No. 26, bifilar.
L_2 (1–4) — 2.0 mh., approx. 80 turns No. 26, bifilar.
L_2 (5–6) — 0 to 20 turns No. 26, single.
L_3 (1–4) — 0.7 mh., approx. 47 turns No. 26, bifilar.
L_3 (1–8) — 2.3 mh., approx. 86 turns No. 26, bifilar.
L_4 (1–4) — 5.0 mh., approx. 125 turns No. 26, bifilar.
L_4 (5–6) — 0 to 20 turns No. 26, single.
L_5 (5–8) — 14.0 mh., 160 turns No. 28, bifilar.
L_5 (1–4) — 1.4 mh., approx. 47 turns No. 26, bifilar.
All wire Formvar or d.s.c. — see text.
L_1 and L_5 wound on Western Electric P476930 core ring.
L_2, L_3 and L_4 on Western Electric P284395 core ring.

$$\text{Approx. turns P476930} = 1000/\sqrt{\frac{164}{L}}$$

$$\text{Approx. turns P284395} = 1000/\sqrt{\frac{79}{L}}$$

Resonant Frequencies

Inductor	*Capacity*	*Freq., kc.*
L_1 (5–8)	C_1	19.1
L_2 (1–4)	C_3	20.5
L_2 (1–6) with C_3 connected	C_2	19.1
L_3 (1–4)	0.1 μfd. ±5%	19.0
L_3 (1–8)	C_4	19.0
L_4 (1–4)	C_5	21.5
L_4 (1–6) with C_5 connected	C_6	18.9
L_5 (1–4)	C_7	18.9

The series arm C_5-C_6-L_4 is considered and modified in the same manner.

This now leaves only four capacitors, C_1, C_2, C_5 and C_7, to be selected to plus or minus 5 per cent. Each capacitor may of course be made up of two or more units in parallel if necessary to obtain the correct value.

The filter may be further modified by the addition of separate windings to L_1 and L_5 to permit operation directly from a ring modulator and into the grids of a balanced tube modulator. This adds little additional cost, and accurate balance can be easily obtained by using bifilar windings.

In the design given, an impedance of 500 ohms was selected for the input winding of L_1, since a copper-oxide ring modulator operates quite satisfactorily into this impedance. The impedance of the output winding of L_5 is a compromise between desired voltage step-up and keeping the number of turns to a value that permits hand winding. The completed filter design after all modifications is shown in Fig. 6. In the event other input or output impedances are desired, the number of turns and method of connection of these added windings may be altered to meet the requirements. Since the impedance varies directly with the inductance of the windings (with 1.4 mh. the inductance for 1000 ohms impedance), the required inductance in millihenrys for any new impedances may be found by dividing the new impedance in ohms by 1000 and multiplying the result by 1.4. The number of turns required can be found from the formulas for the inductors given in Fig. 6.

Filter Alignment

As has previously been mentioned, the *LC* combinations must be resonant at the desired frequency. In an *m*-type filter with closely-spaced rejection frequencies, it is very important to hold to very close frequency tolerances; while a constant error is not serious the spacing of one frequency to the next is critical.

Heretofore, it has been considered necessary to use expensive laboratory equipment, which is out of the reach of many. Signal generators for the range of 15 to 30 kc. are not common, and those available are usually not of sufficient accuracy. However, with the aid of a BC-221 frequency meter the main obstacle has been removed. The fundamental frequency range of the low band of the BC-221 is 125–250 kc., and it has sufficient output voltage to give a reasonable indication on most oscilloscopes. The BC-221 is used only as a calibration means for the test signal generator. The test generator may easily be made from the junk box, and the usual calibration problem solved by the BC-221. In fact, good procedure is to use only a rough calibration and use the BC-221 continuously for frequency set. The method of connection of the frequency-check system is shown in Fig. 7. The oscilloscope vertical and horizontal inputs are used to give the familiar Lissajous figures as a means of comparing frequencies. Since most of the frequencies needed from the test generator are one-tenth that of the BC-221 it is convenient to use the chart calibration points for 125–250 kc. By moving the decimal point one place to the left and obtaining a 10:1 Lissajous pattern on the oscilloscope the frequency may be read directly. Other multiples must, of course, be used for some frequencies.

An *LC*-type signal generator is recommended for best stability, and particularly if one has to be constructed.

A method of resonating that gives accurate results is shown in Fig. 7. This method measures all *LC* combinations as a series-resonant circuit. With the *LC* combination connected as shown, a sharp dip in amplitude occurs when the frequency is at the exact series-resonant point, since the impedance is then a minimum. Although not critical, *R* of Fig. 7 should be the smallest value that will still give a readable indication. When an entire series arm is resonated the dip will not be as great but will be very sharp.

Coil Construction

In selecting inductors for the filter, the *Q* is of primary importance. *Q* values of at least 150 are necessary for all inductors except possibly L_1 and L_5. L_1 and L_5, as in the case of C_1 and C_7, are in parallel with the terminations, and losses here have much less effect. While many types of inductors might be used, the toroidal type has many advantages and core rings of molybdenum Permalloy are now available to the amateur.

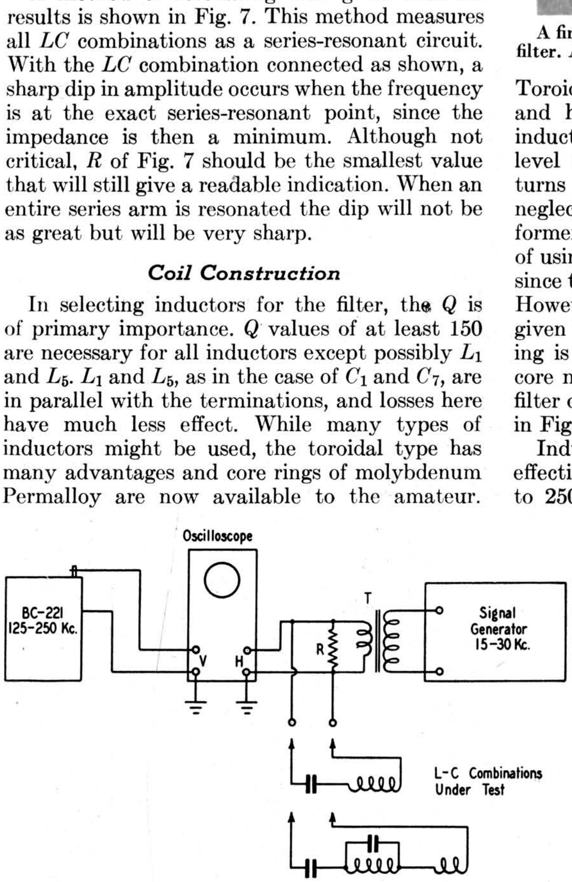
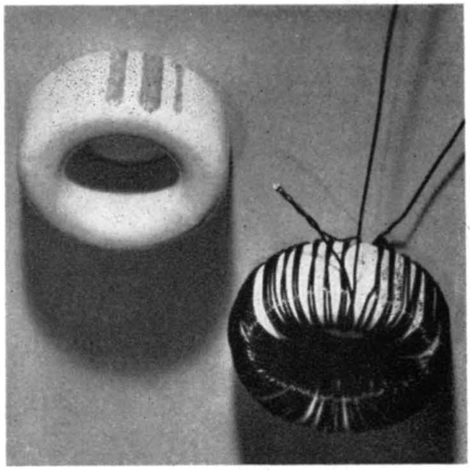


Fig. 7 — **The method used for checking coil-and-condenser combinations. An accurate frequency check is obtained by using a BC-221 to check the 10th harmonic of the test signal generator. The *LC* combination under test is adjusted for minimum horizontal amplitude at the desired frequency.**
R — 1 to 10 ohms, ½ watt. See text.
T — Step-down transformer. A 500-to-6-ohm audio transformer is suitable with generator outputs of 500 ohms or less.



A finished toroid coil of the type used in the sideband filter. An unwound core is shown at the left.

Toroidal coils of this material are small in size and have a very low external field, and the inductance remains quite constant with power level and temperature. The coupling between turns is high, so that leakage reactance may be neglected in the design of the built-in transformers and tapped coils. The one disadvantage of using toroidal coils is the difficulty of winding, since the wire must be threaded through the core. However, in this filter special attention was given to keeping the inductances low, and winding is not too difficult. Two different grades of core material were used in the inductors for the filter of Fig. 6 (attenuation characteristics shown in Fig. 5).

Inductors L_2, L_3 and L_4 use cores having an effective permeability of 60, producing *Q*s of 200 to 250 at 20 kc. L_1 and L_5 cores were of 125 permeability, reducing the required number of turns and still permitting *Q*s of over 100. The construction data in Fig. 6 give the approximate number of turns of the inductors when using Western Electric core rings P476930 for L_1 and L_5, and P284395 for L_2, L_3 and L_4. P476930 and P284395 have nominal inductances of 164 and 79 millihenrys respectively per 1000 turns. The approximate number of turns for a specified inductance, as given by the manufacturer, is found by the formulas given in Fig. 6.

Since a tolerance is allowed on the capacitors, and the permeability of the cores varies slightly, the exact number of turns will vary and must be determined by measurement. For this reason sufficient length of wire should be allowed for the windings so that the additional number of turns necessary may be found by test. The extra length of leads will not affect the test, and these leads may later be cut to proper length after the correct number of turns has been determined.

Wire size is not critical and deviation from that given in Fig. 6 may be made if winding area does not limit. "Formvar" insulation, or the equivalent, is recommended and is easy to wind, but single silk or nylon is satisfactory. Plain enamel not of the Formvar type is to be avoided, because of the possibility of shorted turns.

In order to reduce the number of times the wire must be threaded through the core ring, all windings are bifilar except the adjustment windings *5–6* of L_2 and L_4. In the bifilar type of winding, two wires are held together and wound as one. After winding, the start of one wire (*3*) may be connected to the finish of the other (*2*), thus connecting the two in the series-aiding manner. As in telephone practice, the numbering of

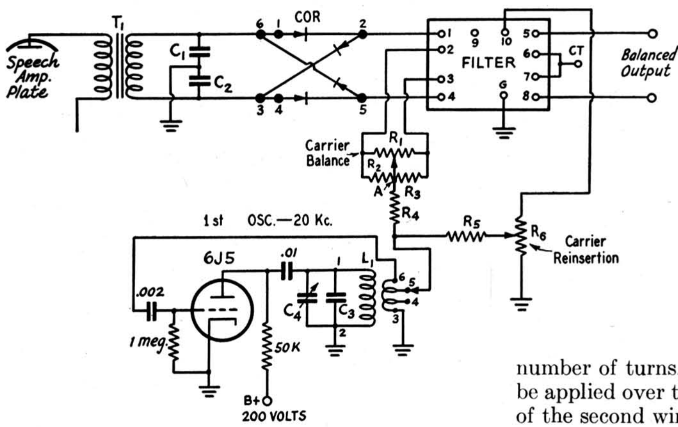


Fig. 8 — **A suggested circuit for the 20-kc. oscillator and balanced modulator to be used with the filter.**

C_1, C_2 — 0.05-μfd. ±20%, matched to within 1% by trial.
C_3 — 0.01-μfd. silver mica.
C_4 — 200-μμfd. variable or adjustable.
R_1 — 5000-ohm potentiometer.
R_2, R_3 — 150 ohms, ½ watt.
R_4 — See text.
R_5 — 3000 to 5000 ohms.
R_6 — 1000-ohm potentiometer.
L_1 — 284 turns or 142 bifilar, No. 26 Formvar or s.s.c., for coil 1–2. Coil 3–6 is 40 turns of same tapped every 10 turns. Both coils are wound on the same W.E. P284395 core ring.
COR — Copper-oxide modulator (Varistor). See text.
T_1 — Output transformer, plate to 500 ohms.

windings connected for series aiding is such that the direction of current at one instant is from *1* to *2*; *3* to *4*; *5* to *6*; etc. Thus if *2* and *3* and *4* and *5* are connected together and external connections are made to *1* and *6*, the windings are series-aiding.

In winding, the length of wire to be pulled through for each turn may be halved by starting at the center of the bifilar (doubled) length of wire in the core ring, and then winding in opposite directions through the core ring respectively with each end of the bifilar wire. The wire should be evenly distributed around the core ring, but this is not particularly critical.

The following procedure for proper identification and labeling of bifilar windings may be used: Select one of the ends of the completed winding and arbitrarily label them 1 and 3. Now by use of an ohmmeter, locate the wire at the opposite end of the winding which checks continuity to "1." This of course will be "2" and may be spliced to "3." With the exception of the input and output winding of L_1 and L_5, the free wires, 1 and 4, may be left long and any additional turns necessary may be obtained by winding on singly, with care that the wire continues through the core in the same direction.

The bifilar windings *5–6* and *7–8* follow the same procedure. However, when two windings are to be series-connected such as *1–2*, *3–4* and *5–6*, *7–8*, care must be taken in selection of the end of winding to label *5*, *7*. The proper labeling is such that the wire ends *5*, *7* pass through the core center in the same direction as the wire ends *1*, *3*.

The input windings of L_1 (*1–4*) and output windings of L_5 (*5–8*) are not critical in inductance and may be wound first to the specified number of turns. If desired, a layer of tape may be applied over these windings before application of the second windings.

L_3 (*1–4*) is wound and resonated with a 0.1-μfd. test capacitor to 19.0 kc. Adjust to the nearest turn that produces resonance closest to the exact frequency. C_1 and C_7 may be paralleled and used temporarily for the test capacitor. The second winding of L_3 (*5–8*) is now applied and series-connected with the inner winding, *1–4*. Turns are adjusted to secure resonance with C_4 at 19 kc. No connection is made to the tap during adjustment. Note that wide tolerances on C_4 are allowed and the exact number of turns of L_3 will depend on this tolerance.

L_1 (*5–8*) and L_5 (*1–4*) are wound and resonated with their associated capacitors, C_1 and C_7.

L_2 (*1–4*) is now wound and resonated with C_3. As previously mentioned, the value of C_3 may vary over a wide range (plus tolerance), and will determine the number of turns of windings *1–4*. Note that the total number of turns for L_2, including adjustment winding *5–6* depends only on the exact value of C_2. Thus if C_3 is large, winding *1–4* will have fewer turns, and *5–6* will have more. After resonating C_3 connect it in parallel with windings *1–4* and the combination in series with

C_2. Check the resonant frequency of the entire series arm, less winding *5–6*. It should be higher than the frequency as given in Fig. 6. Now, wind turns on for the trimming winding *5–6*, and, with it connected, recheck the resonant frequency. Adjust turns of *5–6* until correct frequency is obtained. In the event that the resonant frequency was lower than the correct value before the addition of *5–6* it is an indication that C_3 was too low; and the entire adjust-and-check procedure must be repeated with a larger value for C_3.

L_4 is now wound and resonated with C_5 and C_6 following the same procedure as for L_2, C_3, C_2.

The filter may now be wired temporarily for test before mounting. The method of connection for test is shown in Fig. 5. While a sensitive voltmeter or decibel meter of high impedance is necessary for accurate measurement, an oscilloscope may be used instead for an approximation. If the filter is flat through the passband and attenuates rapidly on the high side it is likely that no errors have been made. If the oscilloscope gain is adjusted for full deflection at the center of the passband, the deflection at points above about 20.4 kc. should be barely visible if at all.

The mounting of the components will be left to the builder, but it is to be noted that the inductors may be mounted very close together and near metal surfaces without harmful effect, with the possible exception that L_1 and L_5 should be given some separation from one another. A metal screw may be used through the center of an inductor without harm provided it does not constitute a shorted turn, as it would if metal washers were used on both sides and the washers connected together.

A suggested schematic using the filter is shown in Fig. 8. The speech amplifier should feature low- and high-frequency cut-off as with any 'phone transmitter. Some high-frequency attenuation may be obtained by the action of the secondary of T_1 with C_1, C_2. It is well first to run a frequency-response check on the speech amplifier including T_1, C_1, C_2 with the modulator disconnected and a 500-ohm resistor substituted.

The 20-kc. oscillator shown uses a toroidal inductor. Other types of oscillators will perform satisfactorily if the output impedance is held low. The number of turns of inductor L_1 and value of C_3 may be adjusted for proper frequency using the BC-221 and the proper feed-back adjusted by the secondary winding *3–4–5–6*. The taps on this winding are desirable to adjust the voltage at the junction of R_2, R_3 from 2 to 5 volts. Selection of 100 to 500 ohms for R_4 also permits some adjustment. R_5 should be as high as possible for least loading on the oscillator and filter, still permitting enough 20-kc. output for any desired amount of carrier reinjection.

One point not obvious is that R_2 and R_3 with R_1 in parallel are actually in series with the input to the filter. The values chosen normally give a good impedance match between the modulator and filter. If 1N34s or vacuum tubes are used instead of copper oxide for the modulator, a resistor may have to be placed across filter terminals *1* and *4*, and R_2 and R_3 lowered in value. Proper match may be noted when audio is fed into the speech amplifier and varied from 200 to 3000 cycles. If the speech amplifier has previously checked flat, the output from the filter at terminals *5* and *8* as measured with a voltmeter or noted by the oscilloscope should vary as the response through the filter alone with frequencies of 19.8 to 17.0 kc. A ripple in output amplitude indicates incorrect modulator match.

Copper-oxide "Varistors" available in surplus, which have proven satisfactory with the values given, are Western Electric D162258, D163139 and D98914. The terminal numbering given for "COR" of Fig. 8 is for these types.

Modulator balance for maximum carrier reduction is normally quite simple. A sensitive voltmeter or oscilloscope is connected to output terminals *5* and *6*. With no input to the speech amplifier and R_6 turned for minimum carrier, adjust the carrier balance control R_1 for minimum output.

Balance should be obtained near the center of the adjustment range. If not, a trimming resistor may be paralleled with R_2 or R_3. Some capacity unbalance in the Varistor or input winding of the filter may prevent sufficient carrier balance and small values of capacity may be added from filter terminal *1* or *4* to ground. Capacity may also be tried across C_1 or C_2.

Note that any hum in the speech amplifier will appear as an output carrier, but of course will be 60 or 120 cycles from the true carrier. Hum may be identified by temporarily shorting the primary of T_1.

Audio is now connected to the input of the speech amplifier and the level adjusted to a maximum of 0.25 volt at the output of T_1.

If the output of the speech amplifier is a pure tone the output of the filter should be a single frequency of 20 kc. minus the audio frequency. Using a sweep rate that is a submultiple of the audio input frequency, a check may be made for the presence of a modulation envelope. Such a trace represents more than one frequency in the output and may be caused by distortion in the speech amplifier or overloading of the modulator. A slight modulation pattern is permissible as this represents only a slight distortion of speech and not spurious signals out of the passband.

The modulator is now ready to be connected to the succeeding stages of the exciter.