SSB, Jr.

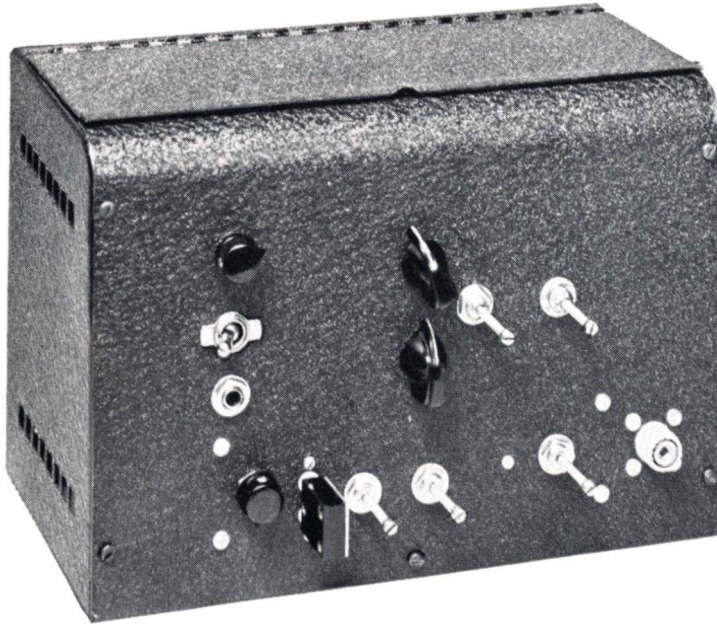## Presenting a 3-Tube 5-Watt SSB Transmitter with Superior Performance

Fig. 1. Front panel view of the SSB Jr. For single-frequency operation none of the controls need be adjusted (except the audio gain control). Front-panel mounting of the controls permits a compact physical layout to be obtained.

FEATURES—

Simple to construct

Uses inexpensive parts

Has sideband-reversing control

Usable as emergency, portable or home transmitter

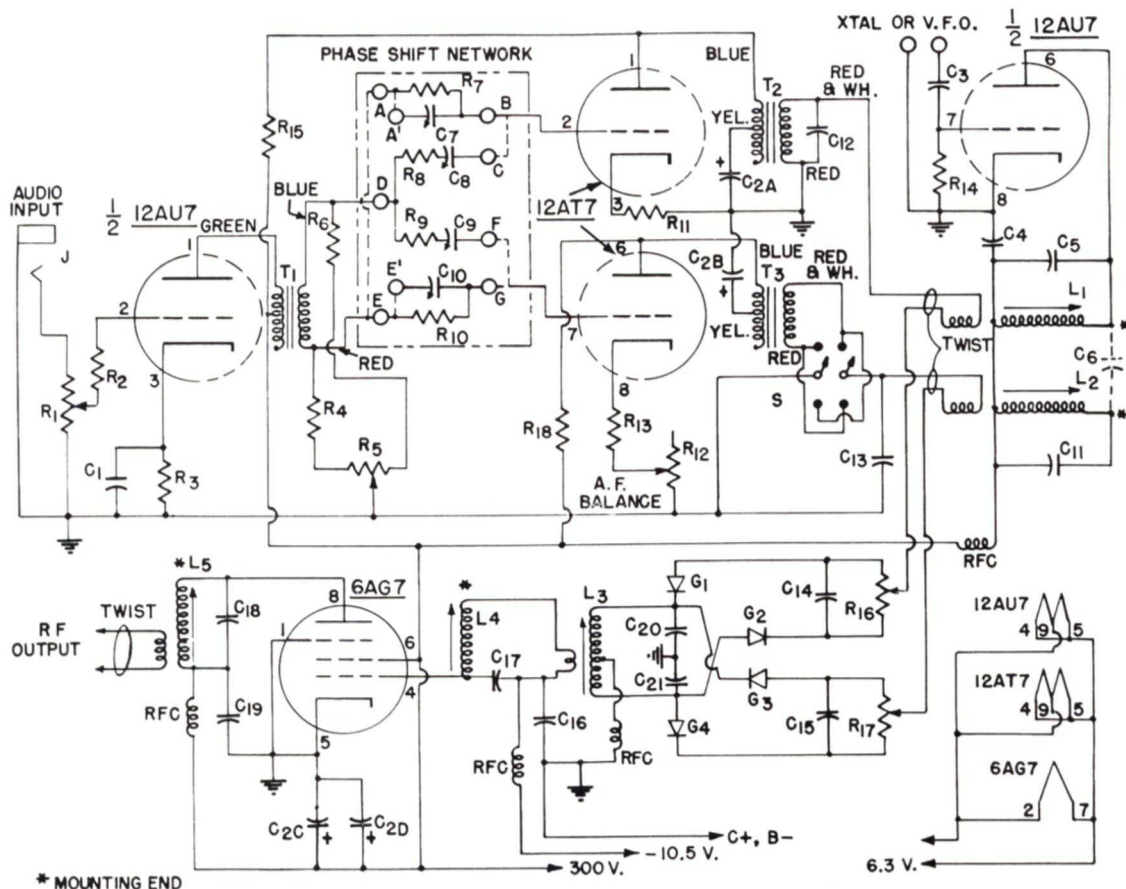## CONTENTS

SSB, Jr.

ELECTRICAL CIRCUIT

Fig. 2. Circuit diagram of the SSB Jr.

Circuit Constants

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

C_1 .... 0.5 mf 200 volt paper
C_2 .... 20-20-20-20 mf 450 volt electrolytic
C_3, C_4, C_{12}, C_{13}, C_{16} .... 1000 mmf mica or ceramic
C_5, C_{11}, C_{17} .... 250 mmf mica ±10%
C_6 .... May not be necessary, see text
C_7 .... 2430 mmf (0.002 mf mica ±5% with 170-780 mmf trimmer in parallel)
C_8 .... 4860 mmf (0.0043 mf mica ±5% with 170-780 mmf trimmer in parallel)
C_9 .... 1215 mmf (0.001 mf mica ±5% with 50-380 mmf trimmer in parallel)
C_{10} .... 607.5 mmf (500 mmf mica ±10% with 9-180 mmf trimmer in parallel)
C_{14}, C_{15} .... 0.005 mf mica or ceramic
C_{18} .... 350 mmf 600 volt mica ±10% (250 mmf in parallel with 100 mmf)
C_{19} .... 0.01 mf mica or ceramic
C_{20}, C_{21} .... 0.002 mf mica ±10%
G_1, G_2, G_3, G_4 .... 1N52 germanium diode or equivalent, see text
J .... Open circuit jack
L_1, L_2 .... 33 turns No. 21 enamel wire close wound on Millen No. 69046 iron core adjustable slug coil form. Link of 6 turns of hookup wire wound on open end.

L_3 .... 16 turns No. 19 enamel wire spaced to fill Millen No. 69046 coil form. Tap at 8 turns. Link of 1 turn at center.
L_4 .... Same as L_1 except no link used
L_5 .... 28 turns No. 19 enamel wire. Link on open end to match load. (4 turn link matches 72 ohm load.)
RFC .... Radio-frequency choke 0.5 millihenry
R_1 .... 1 megohm potentiometer
R_2 .... 10,000 ohm, ½ watt
R_3 .... 750 ohm, ½ watt
R_4 .... 430 ohm, ½ watt (±5%)
R_5 .... 100 ohm potentiometer
R_6 .... 1600 ohm, ½ watt (±5%)
R_7, R_{10} .... 133,300 ohm, ½ watt (±1%)
R_8, R_9 .... 100,000 ohm, ½ watt (±1%)
R_{11} .... 510 ohm, ½ watt (±5%)
R_{12} .... 500 ohm potentiometer
R_{13} .... 330 ohm, ½ watt
R_{14} .... 47,000 ohm, ½ watt
R_{15}, R_{18} .... 20,000 ohm, 1 watt
R_{16}, R_{17} .... 1000 ohm potentiometer
S .... DPDT toggle switch
T_1 .... Stancor A-53C transformer
T_2, T_3 .... UTC R-38A transformer

The **SSB Jr.** is a complete single-sideband transmitter—just add microphone and antenna and you are on the air. No longer must amateurs feel that single-sideband equipment is too complex to understand or too complicated to build. The **SSB Jr.** rig is no more difficult to build or adjust than any modern 3-tube transmitter. This rig should bring **SSB** within the reach of anyone that is interested.

Further, any amateur can build the SSB Jr. rig and be assured that his single-sideband signal will be second to none in quality. Performance has not been sacrificed in the interest of simplification.

The peak power output is 5 watts and the total power input, not including filament power, is 18 watts (300 volts at 60 ma.). The SSB Jr. rig features a self-contained crystal oscillator (or buffer for VFO operation), 40 db. sideband suppression, and mechanical and electrical ruggedness that make it ideally suited as a complete portable, mobile, emergency transmitter, or as an exciter for a home transmitter.

The system used in the generation of the single-sideband signal is a simplified phasing method that is daringly direct and effective. Inexpensive and easily-available components are used throughout.

All of the information necessary to construct and adjust the SSB Jr. appears in this article. Technical details on the new phase-shift network and the new modulator design are explained in the Designer's Corner section of this issue.

With reference to the circuit diagram, Fig. 2, the first tube, a 12AU7, is a twin-triode, combination speech amplifier oscillator. A 12AT7 serves as a twin-channel amplifier in the output of the phase-shift network, and the final is a 6AG7 pentode.

Starting with the audio circuit, an input gain control potentiometer feeds the grid of the self-biased input tube, which is one-half of the 12AU7 miniature tube. The output of this tube is coupled into a newly designed audio phase-shift network by means of transformer T_1. The outputs of the phase-shift network feed separate triode sections of the 12AT7 miniature tube. These two tube sections are transformer coupled to two balanced modulators each of which employs a pair of germanium crystal diodes.

The balanced modulators are also supplied by r-f signals from the crystal oscillator, which is the other half of the 12AU7. These r-f signals are picked up by separate link windings on L_1 and L_2, which comprise portions of a 90 degree r-f phase-shift network in the plate circuit of the oscillator. The balanced modulators work into a balanced load circuit (L_3, C_{20}, C_{21}) which is link coupled to the grid circuit (L_4, C_{17}) of the class AB_1 linear power amplifier tube, a 6AG7.

This power amplifier works into a conventional tank circuit (L_5, C_{18}) that is link coupled to the load. All circuit tuning is accomplished by adjustable slug-tuned coils wound on Millen No. 69046 powdered-iron coil forms.

Sideband switching is accomplished by the reversal of audio polarity in one of the audio channels (switch S). Provision is made for equalization of gain in the audio channels, this equalization being necessary in order to achieve normal sideband cancellation. In addition, a semi-fixed control (R_5) is provided for phase-shift network adjustment. Use of this control eliminates the need for a special transformer, or the need for two non-standard precision resistors. Stable modulator balance is achieved by the balance/buffer resistors R_{15} and R_{17} in conjunction with the germanium diodes.

The audio characteristic of the SSB Jr. is designed to emphasize the intelligence-bearing frequencies from 300 to 3000 cycles per second. This feature is obtained jointly by the action of C_1 and the audio transformer T_1. Low differential phase-shift is maintained in audio circuits following the phase-shift network by means of lightly loaded output transformers which are shunt-fed to reduce harmonic distortion caused by direct current in their windings.

A 5 by 7 by 2 inch chassis provides ample space, with good access, for all component parts. A cabinet, as shown, may be used, although this is not essential. It is recommended that parts layout shown in the sketches and the photographs be followed exactly. Obviously other layouts will work, but the layout shown has been carefully made and many layout problems have been eliminated.

Before starting work on the main chassis it is advisable to make the audio phase-shift network board. This is diagrammed in Fig. 3. The base material may be thin bakelite or any insulating material. The dimensions are 4 inches by $2\frac{1}{8}$ inches. Note that one corner is cut off to permit access to the 12AU7 tube. This board uses four fixed mica condensers which are padded with four adjustable mica trimmers, and four precision resistors (Continental Nobleloy X-$\frac{1}{2}$, plus or minus 1% tolerance). In the unit shown R_8 and R_9 are as specified, that is, they are Continental Nobleloy 100,000 ohm resistors. However, the 133,300 ohm resistors were made by taking two 150,000 ohm precision Continental Nobleloy resistors and paralleling each of them with a one-half watt 1.2 megohm (plus or minus 10% tolerance) resistor. Careful selection of the 1.2 megohm units will permit close adjustment to the desired value of 133,300 ohms. A convenient way to mount the 1.2 megohm resistors is to slip them inside the hollow body of the precision 150,000 ohm resistors.

The phase-shift network sub-assembly is mounted on three half-inch long spacers under the chassis directly below transformers T_1 and T_2. It is best to dress the leads from these transformers flat against the chassis to clear the phase-shift network. Time will be saved by installing the network sub-assembly as the last step in the construction.

Mount the phase-shift network elements as shown in Figs. 3A and 3B. The dashed connections should be omitted initially, since the detailed alignment procedure described later presumes that these connections will be made at the proper time only.
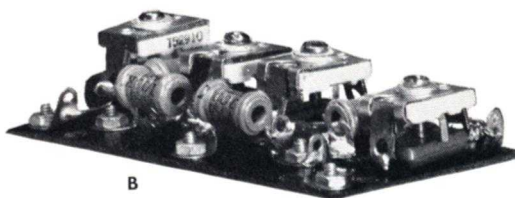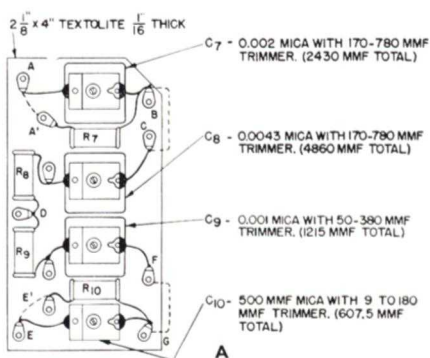

Fig. 3. (A) Mechanical arrangement of the audio phase-shift network. (B) Detail view of the audio phase-shift network

A word of caution about the coils. Make sure that the hot and cold ends are as specified on the circuit diagram—the asterisk indicates the end which is the mounting end, that is, the end with the long tuning screw.

The links on the coils are wound over the cold end, as indicated in Fig. 11. As a suggestion, wind the links with solid insulated hookup wire. This type of wire is convenient, holds on well, and makes a nice looking job. Twist the wires together when running from one coil to another coil, or to another connection point. A small terminal strip may be placed under L_1 to serve as a convenient junction point for the links coming from L_1 and L_2 and going to the balanced modulators.

The small fixed mica tuning condensers that connect across L_1, L_2 and L_3 are mounted on the coil form terminals. The coupling capacitor between L_1 and L_2 (C_6) is shown dotted in the circuit diagram, since the amount of actual capacitance needed at this point will depend on stray coupling effects in the particular unit you build. More information will be given on this later.

Note that the grid connection of the 6AG7 is above the panel from the hot end of L_4 through a hole in the chassis right next to pin number 4 (the grid terminal) of the 6AG7 socket. Direct strapping of terminals 1, 3 and 5 of this socket to the chassis is desirable to ensure stable amplifier operation. Note also that a 2 by 2½ inch brass or aluminum shield is placed between coils L_2 and L_5 below deck.

The unused transformer leads may be cut off close to the winding and forgotten. The secondary windings of T_2 and T_3 have several intermediate taps that are not used. All leads from the three transformers are fed through small rubber grommets in the chassis to circuits on the underside. All, that is, except the secondary leads from T_3 which remain above chassis. Twist these leads together before running them to the sideband reversing switch on the front panel.

Do not ground either heater lead in the chassis, as you may wish to use an a-c heater power supply or perhaps run your automobile engine while transmitting if the rig is used for mobile work.

Ample mounting space for C_1 and R_3 will be found near C_2, the four-section electrolytic condenser. With reference to C_2, one 20 mf section is C_{2A}, another is C_{2B}, etc. The heater leads that run from the 12AU7 may be cabled together with the other leads from T_1, T_2, T_3.

The germanium diodes deserve special care in handling. Do not bend the leads close to the diode unit itself. The diodes are mounted by means of their leads between the coil terminals of L_3 and the appropriate ends of R_{16} and R_{17}. Protect the germanium diodes from heat while soldering by holding the lead with cold pliers between the diode itself and the end where the soldering is taking place. Further, use only as much heat as is necessary to make a good joint.

A four-wire shielded cable brings power from the power supply to the exciter. The shield serves as the negative plate supply lead and should be connected to chassis ground. A male plug at the other end of the cable makes a convenient connection to the power supply.

The power supply is not unusual in any respect. Any source of power supplying 300 volts and 60 mils or more may be used. It is not necessary to use electronic bias either, and a standard battery supplying 10.5 volts may be used for bias.

The power supply used with the SSB Jr. rig pictured is shown in Fig. 7 and the circuit diagram given in Fig. 6. A 5V4-G rectifier tube feeds a single-section filter to supply 300 volts, and a 6H6 tube acts as a bias rectifier to supply 10.5 volts. Resistor R_1 adjusts the bias voltage obtainable.

The main a-c switch is S_1, and the stand-by switch is S_2. Note that resistor R_2 acts as a low resistance bleeder to drop the positive voltage to zero quickly when the rig is turned off. A double-pole switch is employed with the switch arms tied together, as this arrangement gives the effect of a double break contact.

There is nothing critical about the power supply layout, and any arrangement may be used to suit your convenience.

The SSB Jr. rig as designed requires that a high-output microphone circuit be used. A single-button carbon microphone, connected as shown in Fig. 8B is quite adequate, even desirable, if mobile operation is contemplated.

On the other hand, low-level microphones, such as the usual type of crystal or dynamic microphone, may be used if a one-tube preamplifier is provided. A suggested circuit is shown in Fig. 8A. This preamplifier may be built as a separate unit or incorporated into the SSB Jr. rig. Either the preamplifier shown or the single-button carbon mike circuit will provide in excess of the 2 volt (RMS) signal level required as a minimum input signal to the SSB Jr.

As is true with many transmitter designs, there are some component parts used in the SSB Jr. rig that must be chosen carefully. Obviously, the precision resistors specified are important. If precision resistors are not available—although you should try to get them if at all possible—you may use non-precision resistors which have been checked on a good resistance bridge. You may find that these resistors will change value after they have been used for a while, and that is why it is desirable to use precision resistors initially.

The adjustable mica trimmers used in the audio phase-shift network may be any good grade of mica trimmer.—Those actually used are El-Menco mica trimmers—T52910 for the 170 to 780 mmf range; T52510 for the 50 to 380 mmf range; and T52310 for the 9 to 180 mmf range.

Resistors R_4, R_6 and R_{11} are specified as plus or minus 5% tolerance. This is because the values stated are required, and these values only come in the 5% tolerance series.

The germanium diodes are specified as 1N52 diodes. Other types, such as 1N48, 1N51 and 1N63 may be used instead. If possible, select four diodes which have about the same forward resistance. The forward resistance is the low resistance as checked on an ohmmeter. To determine approximately what it is, measure the resistance in one direction, then reverse the leads to the diode and make a second measurement. The two readings should be quite different. The lower resistance is the one of interest. Make this measurement on the four diodes you intend to use to make sure that the forward resistance of any one of the diodes is within ten per cent of the average resistance of the group.

The diodes used in the rig shown measured approximately 250 ohms on a Weston 772 analyzer when the analyzer was set to the RX10 scale. (Dif-
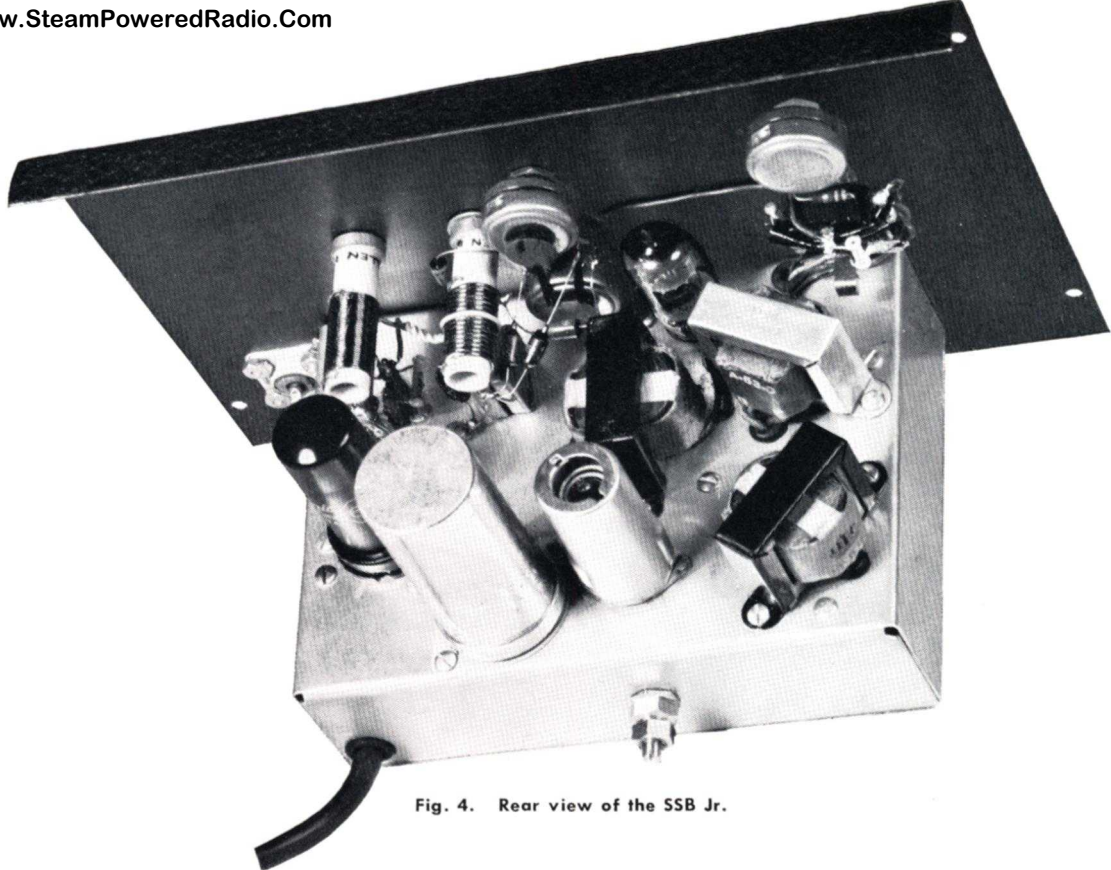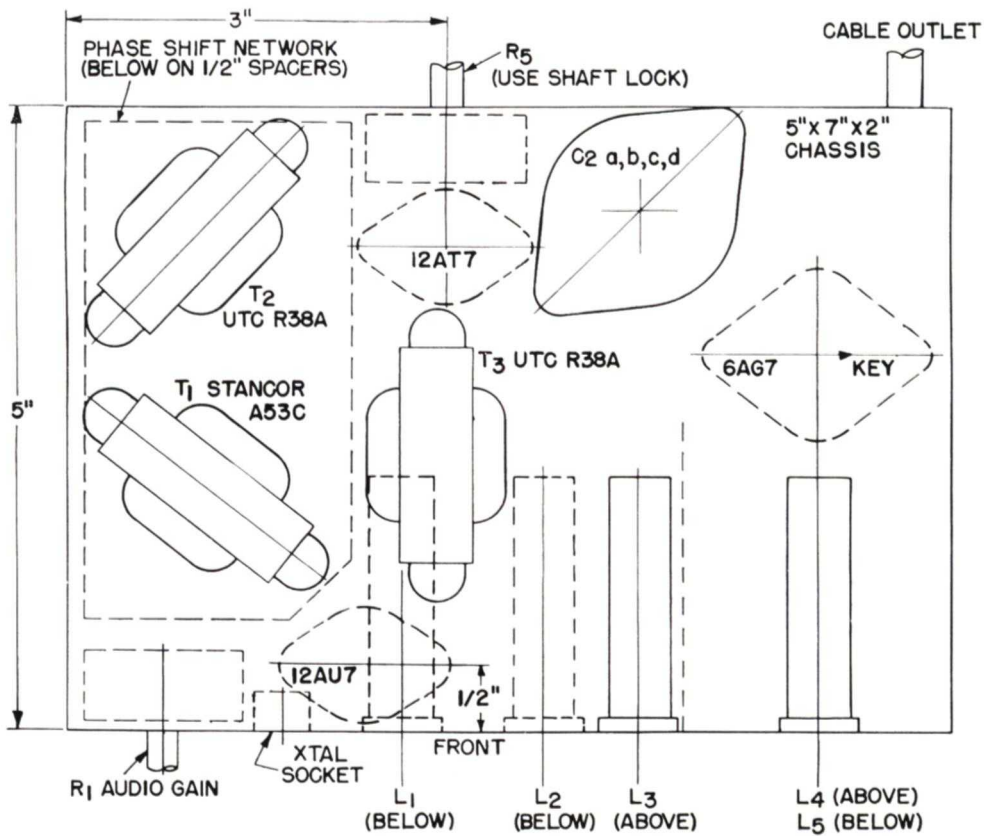
Fig. 4.   Rear view of the SSB Jr.

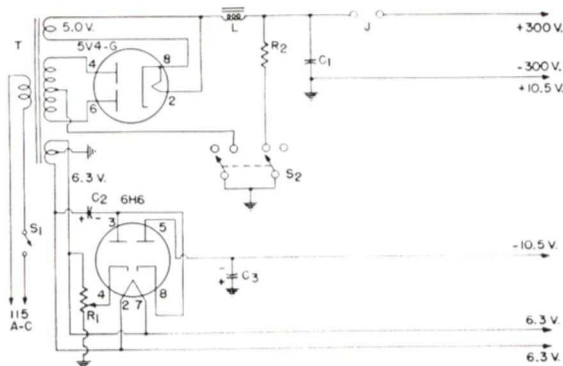Fig. 5.   Chassis layout for the SSB Jr. (top view)

Fig. 6. Circuit diagram of the SSB Jr. power supply

Circuit Constants

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

C₁	40 mf 450 volt electrolytic	R₂	1000 ohm 1 watt
C₂, C₃	50 mf 50 volt electrolytic	S₁	SPST toggle switch
J	Closed circuit jack or terminal posts to permit metering with 0-100 mil meter	S₂	DPDT toggle switch
L	7 henry choke, 160 mils (UTC R-20)	T	Power transformer, 350-0-350 at 75 mils, 6.3 volts at 3 amperes, 5.0 volts at 3 amperes (UTC R-11)
R₁	100 ohm potentiometer		

ferent ohmmeters may give different readings, since the diodes are non-linear in nature.)

It is recommended that transformers T₁, T₂ and T₃ be as specified. Do not make any substitution unless you wish to duplicate a long series of tests to determine if the substitutes are suitable. The types indicated are standard parts, inexpensive, and easy to procure. Observe that the connections are indicated on the circuit diagram by their color code.

It is further recommended that you use Millen No. 69046 coil forms as specified. While the coils are not critical, they must have a certain inductance and distributed capacitance, and if you adhere to the specifications given you should encounter absolutely no difficulty coil-wise.

Initial Circuit Adjustments

The adjustment of the audio phase-shift network circuits is most easily done with the phase-shift sub-assembly out of the chassis. The resistors R₇ and R₈ (and R₁₀ and R₉) should bear the ratio of 133,333 to

Fig. 7. SSB Jr. Power supply

100,000, that is, 4 to 3, as closely as can be determined. If in doubt as to the ratio of the resistors you used, double-check their value on an accurate bridge. The adjustment of the phase-shift network now consists only of setting the four capacitors (C₇ through C₁₀) to their proper values. Several methods can be used. The most accurate will be described.

An audio oscillator capable of operation from 225 to 2750 cycles per second (with good waveform) is required, plus an oscilloscope. The oscillator should be carefully calibrated by the method described later. Connect the output of the audio oscillator through a step-down transformer (the Stancor A-53C will serve nicely) to a 1000 ohm or 2000 ohm potentiometer with the arm grounded.

Adjust the arm position so that equal (but opposite) voltages appear on each half of the potentiometer. A steady audio frequency signal of any convenient frequency may be used with an oscilloscope acting as a convenient voltmeter for this job. Swing the vertical deflection lead from one end of the potentiometer to the other and adjust the arm to obtain equal voltages (a true center tap). Set up a temporary double cathode-follower circuit using a 12AT7 with 500 ohms from each cathode to ground and connect as shown in Fig. 9. (It will be convenient to provide leads M, N, and 1 and 2 with clips at the ends to facilitate checking.) One may use the 12AT7 in the rig as the double cathode follower by temporarily short circuiting the plate of each tube to its respective center tap of the UTC R-38A transformers. Be sure to remove the 12AU7 and the 6AG7 at this time, and of course supply operating voltages for the 12AT7. Pins 3 and 8 should connect to the H and V deflection amplifiers in the oscilloscope, and the oscilloscope common connection should be made to the chassis.

First connect lead M to terminal A on the phase-shift unit, and lead N to terminal A′. Connect leads 1 and 2 to terminal M. (Note that the dashed connections are missing at this stage of adjustment.)
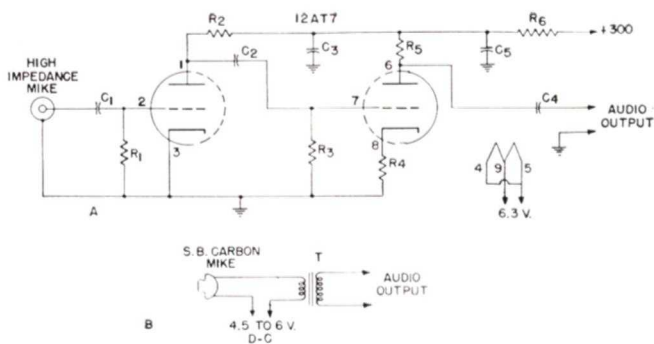
Fig. 8. Suggested microphone circuits for use with the SSB Jr.

Circuit Constants

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

C₁, C₂, C₄ . . . . 0.05 mf 400 volt paper	R₄ . . . . 470 ohm, ½ watt	
C₃, C₅ . . . . 8 mf 450 volt electrolytic	R₅ . . . . 47,000 ohm, ½ watt	
R₁ . . . . 2 megohm, ½ watt	R₆ . . . . 33,000 ohm, ½ watt	
R₂ . . . . 0.1 megohm, 1 watt	T . . . . Microphone to grid transformer	
R₃ . . . . 0.5 megohm, ½ watt		

Adjust the horizontal and vertical gains on the oscilloscope to produce a line about 1½ inches long slanted at 45 degrees when the oscillator is set to a frequency of 490 CPS (an exact method of setting frequency will be described later). If the oscilloscope has negligible internal phase shift the display will be a straight line instead of a narrow slanting ellipse. If the latter display appears it is necessary to correct the oscilloscope phase shift externally by using an adjustable series resistance (a 50,000 ohm potentiometer) mounted at either the vertical or horizontal input terminal, depending on what correction is necessary.

At any rate, the objective here is to get a single straight line at 490 CPS. In some cases a series capacitor may be needed to provide the necessary correction. Try values from 0.05 to 0.0005 mf. Now shift lead 1 from terminal A to terminal B on the phase shifter. Adjust the trimmer of C_7 to obtain a circle on the oscilloscope. It will be noted that as this adjustment is made the display will shift from an ellipse "leaning" to one side through a circle or ellipse (with axes parallel to the deflection axes) to an ellipse which leans the other way. If desired or necessary, the appropriate gain control on the oscilloscope may be changed so that a circle instead of a "right" ellipse is obtained at the point of correct adjustment. After changing the gain control on the oscilloscope, check (and correct, if necessary) the phase shift in the oscilloscope by moving lead 1 back to terminal A, and then repeat the setting of C_7 with lead 1 back on terminal B.

In general, always make certain that the oscilloscope is used in a phase-corrected manner. As a double-check (if the deflection plates in the oscilloscope are skewed, for instance) connect lead 2 to terminal A'. If the circle changes to a slanting ellipse, readjust C_7 to produce an ellipse "half-way" between the ellipse (obtained by switching lead 2) and a circle. Changing lead 2 from A' to A and back again should give equal and opposite skew to the display when C_7 is set correctly. Failure to get symmetrical

ellipses (egg-shaped, or other display) is due to distortion, either in the oscilloscope, the oscillator, the transformer, or the cathode follower. Conduct the test at as low a signal level as possible to avoid distortion.

Next connect leads M and N to terminals E and E', respectively. Connect leads 1 and 2 to E, set the oscillator frequency to 1960 CPS, correct oscilloscope phase shift as before, and move lead 1 to terminal G. Adjust C_{10} for a circle as was done for C_7, using the precautions outlined for that case.

Now connect lead M to terminal D, and lead N to terminal F. Connect leads 1 and 2 to terminal D, set the oscillator frequency at 1307 CPS, correct oscilloscope phase shift as before, and move lead 1 to the junction of R_9 and C_9. Adjust C_9 for a circle on the oscilloscope, as before.

Repeat the above procedure for the remaining R-C pair, R_8 and C_8. Use terminals D and C this time and set the oscillator for 326.7 CPS. This completes except for a final check the adjustment of the phase-shift network. Connect A to A', E to E', B to C, F to G, and A to E. Be certain to remove the temporary short circuiting connections between the 12AT7 plates and T_2, T_3.

If the oscilloscope did not require changes in external compensation over the four frequencies used an over-all frequency check can now be made easily on the phase-shift network. To do this, connect lead 1 to point B, C, lead 2 to point F, G, lead M to point A, A', E, E', and lead N to point D. Now shift the arm of the potentiometer toward M until a circle appears on the oscilloscope screen at a frequency of 250 CPS. Then, as the oscillator frequency is varied from 250 CPS to 2500 CPS, this circle will wobble a little from one side to the other, passing through a perfect circular display at 440, 1225 and 2500 CPS.

The audio band over which the wobble indicates a plus or minus 1.3 degree deviation from 90 degrees is 225 to 2750 CPS, or 12 to 1 in range. This means that when other circuits are properly adjusted, a
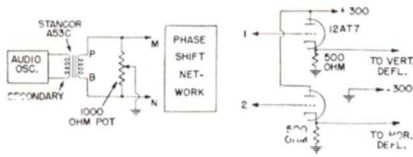
Fig. 9. Audio phase-shift network test layout

sideband suppression ratio of 39 db is possible at the *worst* points within this range. The average suppression ratio will be about 45 db. Proper phase-shift network operation is necessary to obtain this class of performance, so the adjustment procedures have been explained in great detail as an aid toward this goal. The phase shift network should never require readjustment, so that when you are satisfied with the adjustment you may seal the trimmers with cement.

Audio Oscillator Calibration

It will be noted that the frequency ratios are such that the 12th harmonic of 326.7 CPS, the 8th harmonic of 490 CPS and the 3rd harmonic of 1306.7 CPS are all the same as the 2nd harmonic of 1960 CPS, namely, 3920 CPS. Thus, if a stable source of 3920 CPS frequency (such as a thoroughly warm audio oscillator) be used as a reference, the frequency of the test oscillator can be set very closely to one-half, one-third, etc., of this reference frequency if both oscillators feed an oscilloscope and the resulting Lissajous figures observed.

Use of a calibrating frequency in this manner assures that the frequency *ratios* used are correct, even though the exact frequencies used are unknown.

The frequency ratios (just as the resistance ratio previously mentioned) are far more important than the actual values of frequency (or resistance) used.

Transmitter Adjustment

Install the phase-shift network in the chassis, remove the 6AG7 output tube, plug in a crystal (3850 to 4000 KC) or supply a signal to the crystal socket from a VFO at not less than a 10 volt (RMS) level, set L_1 and L_2 for minimum inductance (slug out, counterclockwise) and apply power. The current drain should be about 35 to 40 MA at 300 volts under this condition with the oscillator operating. If the current drain is over 45 MA, turn off the B+ power, adjust L_1, reapply power, etc., until the crystal oscillates. This may be checked by means of a receiver tuned to the crystal frequency. Continue to advance the slug in L_1 with the crystal operating until oscillation ceases. Then back the slug out a few turns to assure stable crystal operation. For VFO input simply adjust L_1 for minimum total current.

Apply an audio signal of 1225 CPS to the input jack of the exciter and connect the horizontal deflection of the oscilloscope to a cathode (pin 3) of the 12AT7, and the vertical deflection to the other cathode (pin 8) after making certain that the oscilloscope is phase-compensated at the frequency of 1225 CPS. Adjust R_5 to produce a circle on the screen. Adjust R_{12} to about mid-range. This test should be made at a reasonably low audio signal level (in general, the lower the better).

Now plug in the 6AG7, after checking to see that a bias of about $10\frac{1}{2}$ volts is supplied. Connect the output link on L_5 to the vertical plates of the oscilloscope (no amplifier used). Deliberately unbalance one of the
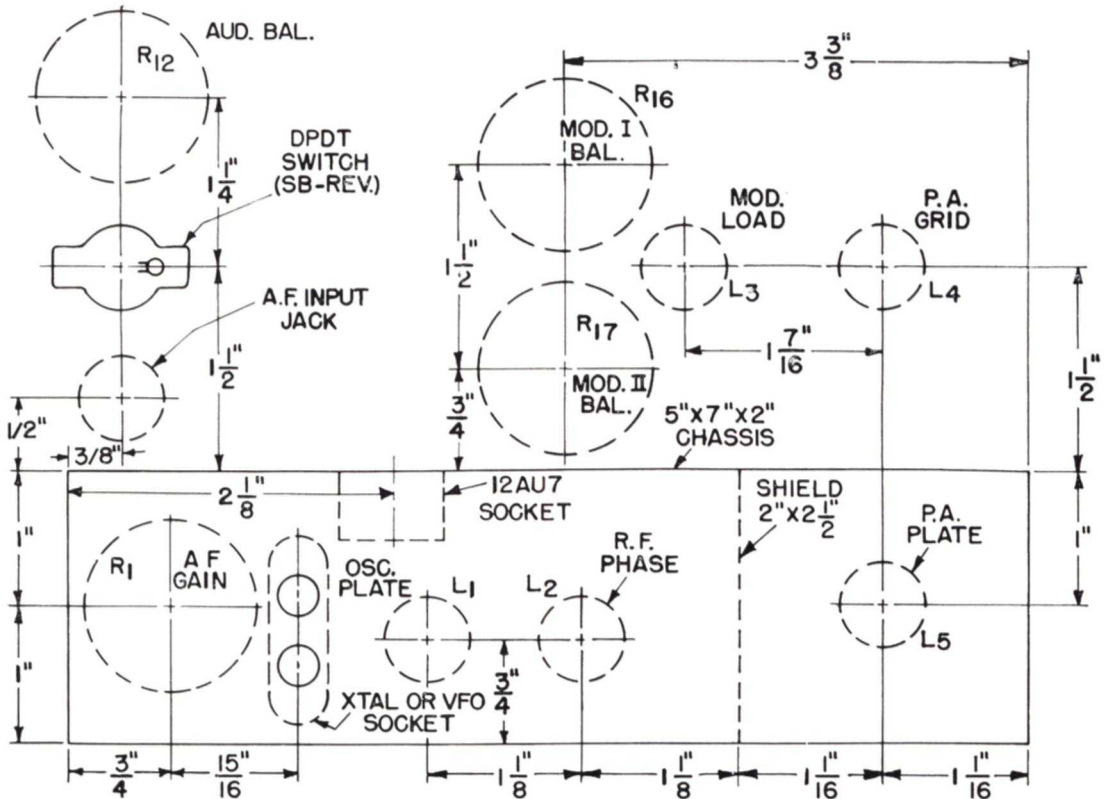

Fig. 10. SSB Jr. panel layout (front view)

modulators by setting R_{16} appreciably off-center. Adjust L_5 for maximum vertical deflection at any convenient sweep speed. This deflection may be small at first since other circuits are not yet tuned. Adjust L_3 for further increase of deflection (maximize), and then finally tune L_4 for maximum output. As this tuning is done it may be necessary to reduce the modulator unbalance to keep from overloading the output stage. Check the tuning again on L_5, L_3, and L_4, in that order. Next remove all audio input by turning R_1 to zero, and, by successive alternate adjustments of R_{16} and R_{17}, balance the modulators for zero output as seen on the oscilloscope. It will be noted that as the correct points are reached the minimum point becomes successively sharper on each control.

Next apply some 1225 CPS audio tone to the exciter by advancing R_1. Undoubtedly some RF envelope will be seen. Adjust L_2 (the RF phase control) in such a direction as to reduce the "modulation" appearing on the output. Remove the tone, check modulator balance (R_{16} and R_{17}), and repeat the adjustment of L_2. The crystal (if used) may stop oscillation during this operation due to interaction between L_2 and L_1 tuning. If so, back out the slug on L_1 until stable crystal operation is obtained. With the 1225 CPS audio signal still applied continue to adjust L_2 for minimum "modulation" or ripple on the envelope, checking modulator balance periodically. When a minimum point is reached, adjust R_{12} to still further reduce this ripple, then adjust L_2 for more reduction, etc. until a substantially ripple-free display is seen.

With L_2 tuned it is now time to check the r-f voltages applied to the modulators. Temporarily remove the audio tone and connect the vertical deflection plate of the oscilloscope to the arm of R_{16}. Always keep the common connection of the oscilloscope grounded to chassis. Note the deflection and then check the voltage on the arm of R_{17} in a similar manner. If this is appreciably lower than the first voltage (on the arm of R_{16}) more coupling capacity (C_6) is necessary between L_1 and L_2.

Usually very little capacity is required, and this can be provided conveniently by making a condenser of two pieces of insulated wire twisted together for half an inch or so. Adjust the amount of capacity by clipping off a little bit at a time to approximately equalize the RF signals appearing on the arms of R_{16} and R_{17}. (Note: check both voltages after each adjustment of capacity, since both voltages will change.) Connect the oscilloscope to read r-f output from L_5 as before, and then check modulator balance. Apply the 1225 CPS tone and make whatever slight adjustment is necessary in L_2 tuning to obtain the ripple-free display obtained before the coupling capacitor (if necessary) was installed. Remove the audio tone and check modulator balance (R_{16} and R_{17}). This completes the adjustment of SSB, Jr. A dummy load may now be connected, or the output used to drive a high power *linear* amplifier.

Operating Information

Note that when changing frequency, L_1, L_2, L_3, L_4 and L_5 should be readjusted, since these circuits constitute the tuning adjustments of the rig. The principal effect of mistuning L_3, L_4, and L_5 will be lower output or efficiency. *The principal effect of mistuning L_2 will be degraded sideband suppression. It is quite important, therefore, to adjust L_2 very carefully.* It may be noticed that when large audio signals are applied, the envelope develops some ripple. There are two possible causes for this action. The first is carrier unbalance (carrier shift), and the other is harmonic distortion in the audio circuits. (It is assumed that a pure sine wave of 1225 CPS is used
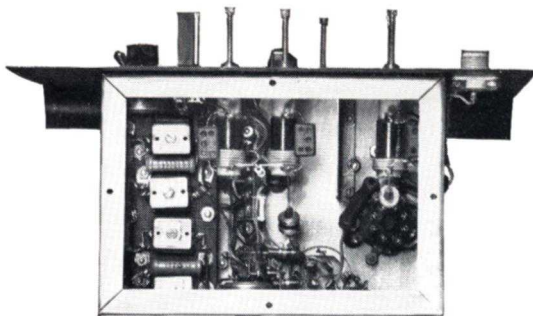

Fig. 11. Under-chassis view of the SSB Jr.

as the input signal.) One may isolate these two effects by setting carrier balance at high-level audio operation (where these effects generally are most pronounced) to reduce the "ripple." With the carrier ripple (which is easily identified when the carrier balance controls R_{16} and R_{17} are moved) balanced out, adjust L_2 slightly (in conjunction with R_{12}) for minimum envelope ripple. The remaining ripple should be less than 5% of the display and is most probably caused by audio distortion, either in the audio source or in the audio system of the transmitter. In observing ripple, the oscilloscope should be synchronized from the 1225 CPS audio signal at a frequency of about 122.5 CPS to show ten cycles or so of carrier ripple. Unwanted sideband ripple will show twice as many peaks, and so will second harmonic audio distortion. Third harmonic audio distortion will show three times as many peaks, etc. Of course, all these distortions (and maladjustments) may occur simultaneously, so a little care and thought is advised. In the sample SSB Jrs. tested, third harmonic audio distortion is the principal component, and is easily identified at high levels.

When feeding a load the total input current will rise to about 80 MA at full level with a single tone input. With speech input the current will rise syllabically from a resting value of about 60 MA to around 70 MA, depending on the waveform. Always use an oscilloscope to determine maximum operating levels. Overload will cause degradation of the sideband suppression, and so is to be avoided. Sideband cancellation adjustments performed at about half peak level are probably the most reliable ones. Carrier balance is best made with little or no audio input. Peak level is the audio signal level which causes flattening of the peaks due to amplifier overload. A higher input level can be used when working into a load, but the overload condition should be avoided while making adjustments and later, too, when operating the rig.

The sideband selector switch is used to control which sideband (upper or lower) is generated. Find out which switch position corresponds to upper sideband by tuning the exciter output signal on a receiver with its BFO supplying carrier. Conduct a talk test and tune the receiver for normal speech output. Then tune the receiver to a slightly *lower* frequency. If the voice pitch *rises*, the upper sideband is being generated. Identify switch positions accordingly.

It takes about 15 minutes from a "cold" start to make all the adjustments described here after a little experience is gained. Do not be frightened away from single-sideband because of a lengthy description of the adjustment procedure, since the adjustments are simple to do, and you will find that the description is actually very detailed and complete. Another reason for not being frightened away from single-sideband is that extremely modest equipment affords the most reliable 'phone communication yet developed.

NOTES ON THE DESIGN OF THE SSB, JR. RIG

> Because the **SSB Jr.** rig design is made possible by a new type of phase-shift network, and a new style modulator, it seemed desirable to have the designer, W2KUJ, explain these units in further detail for the benefit of the technically minded readers of *Ham News*.—Lighthouse Larry

The **SSB Jr.** is a superbly simple rig. Such things just don't happen by accident, however. Throughout the design many new ideas were employed to save space and reduce complication while not sacrificing performance in any way. Easy adjustment for optimum performance was a foremost point of design.

The phase-shift network is an example of simplification of this sort. Literally hundreds of laborious calculations were made along the way to the final solution. The result is a better performing network that has only eight parts and is really very easy to adjust properly. Two methods of adjustment are possible. The first (and preferred one) has already been explained in detail. The other one is obvious. Merely put in accurately measured values and call the job done. The problem here is to obtain the accuracy needed (absolute accuracy) since standards of resistance and capacity are obviously of a different nature. By making adjustments which involve both resistance and capacitance values simultaneously in conjunction with a single reference frequency, almost all sources of error are eliminated. And that is why the preferred method is preferred. All this accuracy is wasted, however, if the components used are not stable enough to hold their values after selection. This is why precision resistors are specified, and why only a small range of adjustment is provided by the trimmer capacitors, since the trimmers are the most likely circuit elements to change. In this way good stability is obtained.

A word about operating conditions necessary for the phase-shift networks. The outputs must feed very high impedance circuits. The effective source impedance should be low, and the voltage supplied to A,E must be *minus* 0.2857 times the voltage supplied to D. Incidentally, the voltage output of each section is equal to the voltage at A,E from zero frequency to a matter of megacycles. The design center frequency for the two networks (yes, there are actually two) is 800 CPS. The differential phase-shift versus frequency curve is symmetrical about this point and holds to within 1.3 degrees from 225 CPS to 2750 CPS, 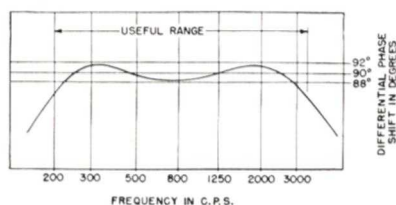as indicated in Fig. 12. A slight error in setting the reference frequency (3960 CPS) will result only in shifting this band up or down by the same percentage. The operating band is adequate—even desirable—for voice communication. One need not fear reports of poor quality when using this rig.

Another simplification which deserves comment is the balanced modulator used in SSB Jr. Let's take a few moments to consider what takes place in the circuit. Fig. 13 shows just one modulator consisting of two germanium diodes, G_1 and G_2 with associated circuits. First, suppose a high frequency signal of a few volts is applied at point R. On the positive crest of signal, current passes through G_2 into the center tapped resonant circuit and tends to pull point S in the same direction. Point T naturally tends to go negative because of the phase inverting properties of the resonant circuit, but, of course, no current flows through G_1. One half cycle later current passes through G_1 from the source, tending to pull point T in the negative direction. But at this time point T would be at a positive potential because of the "inertia" of the resonant circuit. The net result of the battle between G_1 and G_2 to cause current to flow in the resonant circuit is a draw. No net voltage appears across this circuit at the source frequency and energy is dissipated in the balancing resistor and in G_1 and G_2. Thus far, we have currents in the resonant circuit, but none at its operating frequency. This seems like a long way to go to get nothing, but wait.

Now, let us imagine a bias applied at U. If the voltage at U is positive, G_2 will pass more current into the resonant circuit, and G_1 will pass less current. This, in effect, unbalances the circuit and a radio frequency voltage will appear across the resonant circuit, with point S in phase with the voltage at R. If the bias voltage at U is negative, G_1 passes more current than G_2, and the circuit is unbalanced in the other direction. Under this condition the voltage at T will be in phase with that at R. Obviously, if the voltage at U is an audio frequency voltage, the circuit is unbalanced in one direction or the other (at an audio frequency rate) and the resulting radio frequency voltage across the resonant circuit is actually two sets of sidebands with no carrier. When another pair of diodes (such as G_3 and G_4 of Fig. 2) is connected to feed currents into the resonant circuit from related audio frequency and radio frequency sources respectively 90° out of phase with the first, sideband currents caused by these signals flow through the resonant circuit in such a manner as to reinforce one set of sidebands and to cancel the other set. The result is a single-sideband suppressed carrier signal. In the case of SSB Jr., it is a really high grade one.

The function of the balancing resistors (R_{16} and R_{17} of Fig. 2) is to equalize minor differences in the characteristics of the diodes and to balance out stray couplings. Thus, any one balanced modulator is not necessarily perfectly balanced, but the action of two such modulators fed with polyphase signals allows a complete composite balance.

What about operating SSB Jr. in other amateur bands or at other frequencies, in general? As described, the radio frequency circuit design is for the 75 meter

Fig. 12. Audio phase-shift network performance
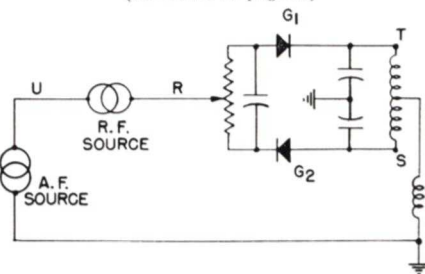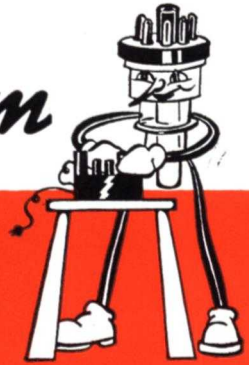

Fig. 13. SSB Jr. modulator circuit

Sweeping the Spectrum

Well, the problem my boss had about the bound volume of *Ham News* is well on its way to being solved. At the time this column was written over two hundred of you had sent in "yes" votes, and when I presented this evidence to the boss he agreed that maybe a bound volume was a good idea. So, if everything goes along as expected I'll be notifying you soon that the bound volume is available. Those of you who sent in "yes" votes will be notified personally.

☒ ☒ ☒

I recently had the pleasure of addressing the Evansville-Owensboro Section of the Institute of Radio Engineers. My subject was the SSB Jr. rig described in this issue of the *Ham News*. D. E. Norgaard, W2KUJ, had also been invited to talk to this group, but he was unable to appear, so I made a wire recording of Don's talk and took it along with me.

Whenever Don or I give talks on single-sideband we like to demonstrate inverted speech, because it is so easy to produce with SSB equipment. As you know inverted speech is that strange sounding stuff that you hear on the short-wave bands on transoceanic communication systems. At least, inverted speech used to be used a great deal, although now more complicated systems of scrambling are employed.

At any rate, you produce inverted speech by taking an upper sideband, let us say, and placing it on the low frequency side of a carrier. This can be done on a receiver by tuning it on the high frequency side of a so-called upper sideband. The effect is to make low pitched sounds high in pitch and vice versa. You should hear the wolf-whistle coming through on inverted speech! I can guarantee that you would never recognize it.

In fact, until you become familiar with inverted speech it is practically impossible to recognize anything. For example, if you say "General Electric Company" into an inverted speech system, what comes out sounds like "Gwunree Oyucktruck Krinkino." Conversely, if you say the latter phrase into an inverted speech system, what comes out sounds like "General Electric Company."

In other words, you can form a new language, and if you speak this new language into an inverted speech system, what comes out is understandable English. As an example, "metz pee wee" means "nuts to you" and "eee wye" says "oh yeah." But you can go even further, as Don and I did. We decided that it would be nice to be able to recite the poem *Mary Had A Little Lamb* in inverted speech, and after an hour of intense concentration we succeeded in the decoding job.

We thought you would like to see this poem in "Sweeping the Spectrum," so here it is:

> Naarow hod O yutty yarng,
> Uts feeious yiz yelt uz snee,
> Arnd I view hair bop naarow yump,
> No yarng yiz sla pee bay.

A word of caution. When practising this poem in inverted speech language, make sure that you are alone. People have strange enough ideas of amateurs as it is!

☒ ☒ ☒

It doesn't take an editor of a magazine long to realize that he has a bunch of sharp-eyed readers. Even though I do know this, every so often something happens that makes me realize that the *Ham News* readers are product-conscious. For example, VE1WM had a question in the September-October, 1950 *Ham News* (page 5) regarding radio interference from fluorescent lamps. I answered the question at some length, and referred to a home-made filter which might be made, consisting of three 0.07 mf condensers connected in delta.

Just the other day one of my readers wrote me, and pointed out that two manufacturing concerns make just such a special condenser, that is, a single unit which contains three 0.07 mf delta-connected condensers. One such concern is Sprague Electric Company, and their interference filter has the number IF-37.

This same person continues, and points out, that the other company making such filters is the General Electric Company! Oh well, looks like I'll have to surround myself with more G-E catalogs. The G-E unit, by the way, carries the number 25F214.

☒ ☒ ☒

The reason this issue of the G-E *Ham News* feels thicker or heavier is that it contains twelve pages. This is not going to be the standard size of the *Ham News* from now on, but in this case, because of the SSB Jr. rig, I deemed it desirable to give as much information as possible on this newest of amateur developments. It is possible that one or two issues a year may be twelve pages long, if the material warrants it.

—Lighthouse Larry

GENERAL DESCRIPTION

Principal Application: The 6AG7 is a metal high-vacuum type power amplifier pentode designed for use in the output stage of television video amplifiers.

Cathode:................Coated Unipotential
Heater Voltage (A-C or D-C).......6.3 Volts
Heater Current................0.65 Ampere
Envelope:................Metal Shell, MT-8
Base:........B8-21 Small Wafer Octal 8-Pin
Base Material:....................Phenolic

The tube is capable of operating at high plate current levels and features a high transconductance.

Direct Interelectrode Capacitances:

Grid to Plate........	0.06	$\mu\mu f$
Input................	13	$\mu\mu f$
Output...............	7.5	$\mu\mu f$
Grid to Screen (Approx).......	5.8	$\mu\mu f$
Grid to Cathode (Approx).......	5.2	$\mu\mu f$
Heater to Cathode (Approx).....	10.7	$\mu\mu f$

PHYSICAL DIMENSIONS

RMA 8-6
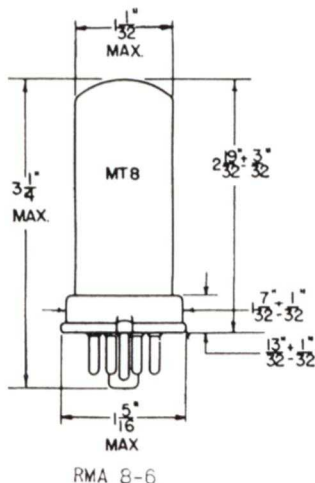
DESIGNER'S CORNER (Cont.)

band, 3850 to 4000 KC. There is no reason, however, to think that equally successful performance would not be obtained on 20 or 10, or even on what is left of 160. It's simply a matter of coil design.

The unit pictured in this issue of *Ham News* was the second one ever built. Ten minutes after the last solder joint had cooled down, the rig was perfectly adjusted and was delivering 5 watts peak power to a 75 ohm dummy load—and I followed the adjustment procedures described in the article. Maybe it will take some people a little longer to read the instructions than it did for me (after all, I wrote them), but 1, 2, 3 procedure really does the job. I didn't peek ahead in the instructions, either.

If you get one-tenth the fun out of building and operating SSB Jr. as I did in designing, building and using it, you are in for the most enjoyment you have ever had in ham radio.—W2KUJ

FROM: