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
Suitable as emergency, portable or home transmitter

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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 simplicity.

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 ohm sideband suppression, and mechanical and electrical ruggedness that makes it ideally suited as a complete portable, mobile, emergency transmitter, or as an exciting for a home transmitters.

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

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

\[ \text{Diagram Description} \]

With reference to the circuit diagram, Fig. 2, the first tube, a 1A177, is a twin-triode, combination speech amplifier modulator. A 1A177 serves as a two-channel amplifier in the output of the phase-shift network, and the final is a 6AG7 pentode.

Starting with the audio circuit, an input gan control potentiometer offers the grid of the self-biased, input tube, which serves as the 1A177 miniature tube. The output of this tube is connected into a newly designed audio phase-shift network by means of transformer T1. The outputs of the phase-shift network feed separate triode sections of the 1A177 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 1A177. These r-f signals are picked up by separate link windings on Lc and Ld, which comprise the r-f phase-shift network in the plate circuit of the oscillator. The balanced modulators work into a balanced load circuit (Lb, Cb, Cg) which is linked to the grid circuit (Lc, Ld, Cc) through the class AB, linear power amplifier tube, a 6AG7.

This power modulator works in the conventional tank circuit (Lt, Ct) that is linked to the load. The circuit tuning is accomplished by simply tuning a slug-tuned coil wound on Millen No. 66016 powdered-iron cores.

Bilateral switching is accomplished by the reversal of audio polarity in one of the audio channels (phase 5). Provision is made for equalization of gain in the audio channels, this equalization being necessary to the extent of obtaining normal sideband cancellation. In addition, a semi-fixed control (R3) 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 use of matching power transformers 7K and 7K2 in conjunction with the germanium diodes.

The audio characteristic of the SSB Jr. rig is designed to offer adequate selectivity over intelligence-band frequencies from 300 to 3000 cycles per second. This feature is obtained by the use of the self-balanced audio transformer T2. Low differential phase-shift is maintained in audio circuits following the phase-shift network by means of lightly loaded output transformers which are terminated to the transformer leads by the inter-stage 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 shows 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 inches. Note that one corner is cut off to permit access to the 1A177 tube. This board feeds two fixed bias condensers which are wired with two adjustable bias voltage trimmers, and two precision resistors (Contingential No.50C-5, plus or minus 1%, tolerance). In the unit shown R1 and R2 are specified, that is, they are Contingential No.50C-500 ohm resistors. However, the 133,300 ohm resistors were made by two 150,000 ohm precision Contingential No.50C-500 ohm resistors and paralleling each of them with a one-half watt 1,3 megohm (plus or minus 10%) tolerance resistor. Careful selection of the 1,3 megohm resistors will permit close adjustment to the desired value of 133,300 ohms. A convenient way to mount the 1,3 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 for transformers T1 and T3. It is best to choose 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. 5A and 6B. The dashed connections should be omitted initially, since the detailed alignment procedure described later assumes that these connections will be made at the proper time only.

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Fig. 3. (A) Mechanical arrangement of the audio phase-shift network. (B) Details of the main phase-shift network.
A word of caution about the coils. Make sure that the kit and cold ends are as specified on the circuit diagram—the numbers indicate the end which is the mounting end, that is, the end with the long terminal nine.

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 conveniently held on, and makes 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 between L2, to serve as a convenient junction point for the links coming from L1 and L2 and going to the balanced modulators.

The small feed mica timing condensers that connect across L1, L2, and L3 are mounted on the coil form terminals. The coupling capacitor between L1 and L2 (C2) is shown dotted in the 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 L4 through a hole in the chassis right next to pin number 4 (the grid terminal) of the 6AG7 socket. Direct straining to terminals 1, 3, and 5 of this socket to the chassis is desirable to ensure stable amplifier operation. Note also that a 2 by 2-1/4 inch brass or aluminum shield is placed between coils L1 and L3, below deck.

The unusual transformer leads may be cut off from the winding and forgotten. The secondary windings of T1 and T2 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 T2 which remain above chassis. Twist these leads together before running them to theabinet 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.

Mounting space for L1 and L3 will be found next to the socket at the rear. The condenser electrolytic condenser. With reference to C5, 20 ml sec is C6, another is C7, and the final 50 ml sec is C8. These capacitors may be called together with the other leads from L3 (see Fig. 11).

The germanium diodes deserve special care in handling. The do not handle the leads close to the diode unit itself. The diodes are mounted by means of their leads between the coil terminals of L4 and the appropriate ends of R1 and R2. Protect the germanium diode from heat while soldering by holding the leads up to the soldering area when the diode is in position. There is no way to determine if a diode is working, consequently the only way to test is to see where the soldering is taking place. Further, use only as much heat as necessary to make a good joint.

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

The power supply used with the SSR Jr. rig is pictured in Fig. 7 and the circuit diagram given in Fig. 6. A 174-6 rectifier tube feeds a single section filter to supply 300 volts, and a 655 tube acts as a boost rectifier to supply 105 volts. Resistor R7 adjusts the bias voltage obtainable.

The main on-off switch is S1, and the stand-by switch is S2, Note. That resistor R17 acts as a low resistance shunt to drop the proper voltage to zero quiescent when the rig is turned off. A double-pole switch is employed with the switch arm tied together, as this arrangement gives the effect of a double break connection.

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

Microphone Considerations

The SSR Jr. rig is designed to require a high-output microphone circuit to be used. A single-button microphone, connected as shown in Fig. 4B 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 microphones, 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 SSR Jr. rig, either the preamplifier shown or the single-button carbon microphone will provide in excess of the 2 volt (RMS) signal level required as a minimum input signal to the SSR Jr.

Component Parts

As is true with many transmitter designs, there are some component parts used in the SSR Jr. rig that must be chosen carefully. Obviously, there are precision resistors specified. It is important that resistors be available—although you should try to get them if at all possible—you may use non-precision resistors which have been clipped on a good resistance bridge. You may find that these resistors are available at a cheaper value and have been used for a while, and that it is desirable to introduce a clip resistor initially.

The adjustable mica trimmers used in the audio phase shift network may be any good grade of mica trimmer. Those actually used are E15-M-bone mica trimmers—T25100 for the 170 to 280 kHz range; T25120 for the 30 to 380 mhz range; and T25310 for the 9 to 100 mhz range. Resistor R5, R8, and R9 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 specified are IN202 diodes. Other types, such as 1N198, 1N519, and 1N63 may be used instead. If possible, select four germanium 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 higher resistance is of no interest. Make this measurement on the four diodes you intend to use. The reverse resistance of any one of the diodes is within ten percent of the average resistance of the group.

The diodes used in the rig shown measured approximately 130 ohms on a Weston 775 analyzer when the analyzer was set to the RX10 scale. (Diff-
Circuit Components
(All resistors and capacitors ±20%; reference values specified otherwise)

- R1, R2, R3, R4, R5: 10k ohm
- C1, C2, C3: 10.0uF

1. a and 125-volt rectifiers
2. Post-it notes on circuit parts to prevent scratching with 0-100 volt meter
3. Heavy chokes
4. 25 volt 1000 uH transformers
5. 2 x 6.8 volt capacitors
6. 125-volt rectifiers
7. Power transformer: 350-500 volts 78 watts
8. 110 volt 1000 uH transformers
9. 2 x 6.8 volt capacitors
10. 125-volt rectifiers

Initial Circuit Adjustments

The adjustment of the audio phase-shift network circuits is most easily done with the phase-shift subassembly out of the chassis. The resistors \( R_1 \) and \( R_2 \) (and \( R_3 \) and \( R_4 \)) should be the ratio of 130,333 to 100,000, that is, 4 to 3, as closely as can be determined. If in doubt as to the ratio of the resistors you use, double-check their values on an accurate bridge. The adjustment of the phase-shift network now consists only of setting the four capacitors (\( C_1 \) through \( C_4 \)) to their proper values. Several methods can be used. This most accurate will be described.

An audio oscillator capable of operation from 250 to 4750 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 Bannor A-53C will serve nicely) to a 1000 ohm or 2000 ohm potentiometer with the arm grounded.

Adjust the trimmer 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. Swapping the vertical deflection lead from one end of the potentiometer to the other and adjust the trimmer to obtain equal voltages (at a true center tap). Set up a temporary double cathode-follower circuit using a 12AT7 with 100 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 M-35A transformers. Be sure to remove the 12AU7 and the SAG7 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 oscillo-
scope 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 con-
nections are missing at this stage of adjustment.)
Adjust the horizontal and vertical gains on the oscilloscope to produce a line about 0.125 inches long 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 of straight lines may appear, with different relative phases, due to temporary gain errors. Try values from 0.03 to 0.006 mfd. Now add lead 1 from terminal A to terminal B on the phase shifter. Adjust the trimmer of C5 to obtain a circle on the oscilloscope. It will be noted that on this adjustment 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 turns 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 C5 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 C5 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 C5 is set correctly. Failure to get symmetrical ellipses (top-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 B and E' respectively. Connect leads 1 and 2 to R, set the oscillator frequency to 1960 CPS, correct oscilloscope phase shift as before, and move lead 1 to terminal C. Adjust C5 for a circle as was done for C4, using the precautions outlined for that case.

Now connect lead M to terminal D, and lead N to terminal E. Connect leads 1 and 2 to R, set the oscillator frequency at 1497 CPS, correct oscilloscope phase shift as before, and move lead 1 to the junction of M and C5. Adjust C5 for a circle on the oscilloscope, as before.

Repeat the above procedure for the remaining B-C pair, M and C5. Use terminals D and C this time and set the oscilloscope for 236 CPS. This completes the test for a final check of the adjustment of the phase-shifting 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 T1, T2.

If the oscilloscope did not require changes in external compensation over the four frequencies used an all-over frequency check can now be made easily on the phase-shifting network. To do this, connect lead 1 to point B, C, lead 2 to point G, C, 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 230 CPS. Then, as the oscillator frequency is varied from 210 CPS to 2500 CPS, this circle will wiggle a little from one side to the other, passing through a perfect circular display at 440, 1250 and 2500 CPS.

The audio band over which the wattmeter indicates a plus or minus 1.5 degree deviation from 90 degrees is 225 to 3750 CPS, or 12 to 1 in range. This means that when other circuits are properly adjusted, 7
sidetone suppression ratio of 38 db is possible at the useful points within this range. The average suppression (ratio) will be about 43 db. Proper phase-shift network operation is necessary to obtain this class of performance, so the adjustment procedure has been explained in great detail so as not to overlook this point. The phase-shift network should never require readjustment, so that when you are satisfied with the adjustment you may seal the trimmers with cement.

Phase-Shift Network Adjustment

It will be noted that the frequency ratios are such that the 12th harmonic of 337 C/Fs, the 8th harmonic of 416 C/Fs and the 3rd harmonic of 1050 C/Fs are all the same as the 3rd harmonic of 905 C/Fs, namely, 3015 C/Fs. Thus, if a stable source of 3015 C/Fs frequency (such as a thoroughly warm audio oscillator) is used as a reference, the generator of the test oscillator can be set very close to the upper limit of the 12th harmonic, the 8th harmonic, and the 3rd harmonic, respectively, frequency if both oscillators are on a stable circuit and the resulting waveform is compared.

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

Phase-Shifter Adjustment

Install the phase-shift network in the chassis, remove the SAGT output tube, plug in a crystal (3800 to 4000 KIC) or supply a signal by the crystal socket from a VFO at not less than 10 volts (3500) level, set L1 and L2 for minimum inductance (plug out, counter-clockwise) and apply power. The current drain should be about 33 to 40 MA at 300 volts under condition with the oscillator operating. If the current drain is over 45 MA, turn off the B+ power, adjust L1, supply power, etc., until the current stabilizes. This may be checked by means of a receiver tuned to the crystal frequency. Continue to advance the slug in L1 with the crystal operating until oscillation ceases. Then back the slug until a few turns to assure optimum operation. Finally, adjust L2 for minimum total output.

Apply an audio signal of 1212 C/Fs to the input jack of the receiver and connect the horizontal (green) lead of the 12AT7, and the vertical deflection to the upper cathode lead of the 12AT7, after making certain that the cathode is phase-compensated at the frequency of 1212 C/Fs. Adjust R8, to about mid-range. This test should be made at a reasonably low audio signal level (5000 volts, the lower the better).

Now plug the SAGT, after checking to see that a bias of about 12 volts is supplied. Connect the output lead on L3, to the vertical plates of the cathode (no amplifier used). Deliberately unbalance one of the
modulators by setting \( R_0 \) appropriately off-center. Adjust \( L_0 \) for maximum vertical deflection at any convenient sweep speed. This deflection may be small at first since other circuits are not yet tuned. Adjust \( L_0 \) for further increase of deflection (maximum), and then finally \( L_0 \) for maximum output. As this tuning is done it may be necessary to readjust the modulator balance to keep from overloading the output stage. Check the tuning again on \( L_0 \) and \( L_1 \) in that order. Next remove all audio input by tuning \( R_0 \) to zero, and, by successive alternate adjustments of \( R_0 \) and \( R_1 \), balance the modulators for zero output as seen on the oscilloscope. It will be noted that at 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 \( R_0 \). Undoubtedly some RF envelope will be seen. Adjust \( L_0 \) (the RF phase control) in such a direction as to reduce the "modulation" appearing on the output. Remove the tone, check modulator balance (\( R_0 \) and \( R_1 \)), and repeat the adjustment of \( L_0 \). The crystal (if used) may stop oscillation during this operation due to interaction between \( L_0 \) and \( L_1 \). 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 \( L_0 \) for minimum "modulation" or ripple on the envelope, checking modulator balance periodically. When a minimum point is reached, adjust \( R_0 \) to still further reduce this ripple, then adjust \( L_1 \) for more reduction, etc. until a substantially ripple-free display is seen. 

It is assumed in this case that the RF voltage is low enough so that the cathode resistor of the RF amplifier may be omitted. This will increase the effective plate load of the RF amplifier, and will decrease the plate current. The RF signal is then applied to the modulators. Temporarily remove the audio tone and connect the vertical deflection plate of the oscilloscope to the arm of \( R_0 \). Always keep the common connection of the oscilloscope grounded to chassis. Note the deflection and then check the voltage on the arm of \( R_0 \) in a similar manner. If this is appreciably lower than the test voltage (as the arm of \( R_0 \) more coupling capacity (C2) is necessary between \( L_0 \) and \( L_1 \) is used. 

Very little coupling is required, and this can be provided conveniently by making a condenser of any order desired, and then connecting it in place of the 0.01 C used. This trick both saves space and adds an adjustment of capacity, since both voltages will be proportional to the band width of the oscilloscope. It will be seen from \( L_0 \) above, and before, and then check modulator balance. Apply audio again and make another trial; if the adjustment is not satisfactory, the adjustment of \( L_0 \) may be made. The last condition should be checked, for the output used to drive the high power linear amplifier. 

Operating information

Note that when changing frequency, \( L_0 \), \( L_0 \) or \( L_0 \) are to be changed, the circuits constitute the tuning adjustments of the rig. The principal effect of changing \( L_0 \) is to produce an equivalent change in the lower output or efficiency. The principal effect of changing \( L_0 \) is that of a degenerative feedback network. It is quite important, therefore, to adjust \( L_0 \) very carefully, since any change in it will cause changes in the output or efficiency of the RF amplifier. Because of the high audio frequency at which the audio signals are applied, the envelope develops some ripple. There are two possible causes for this action. The first is carrier balance (carrier shift), and the other is harmonic distortion in the audio. It is assumed that a pure sine wave of 1225 CPS is used as the input signal.) One may isolate these two effects by setting center balance at high level audio operation (where these effects are generally the most pronounced). 

If center balance is obtained, this makes it easy to determine the carrier ripple (which is easily identified when the carrier balance controls \( R_0 \) and \( R_2 \), or normal balance controls \( R_0 \), or minimum envelope ripple controls \( R_0 \)). The remaining ripple should be less than 2% of the display and is most probably caused by audio distortion, either in the audio source or in the audio output of the transmitter. In observing ripple, the oscilloscope screen should be synchronized from the 1225 CPS audio signal at a frequency of about 1225 CPS to show tous cycles or so of carrier ripple. Unsaturated sideband ripple will show twice as many peaks, and so will spread harmonic audio distortion. Third harmonic audio distortion will show three times as many peaks, etc. Of course, all these distortions (and modulations) may occur simultaneously, so a little care and thought is advised. In the sample SSB Jr. 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 explosively 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 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. 

The sideband switch is used to control which sideband (upper or lower) is generated. Find pin which connectsHRESULT omitted to upper or lower sideband by tuning the exciter output signal on a receiver with its BFO'ing carrier. Connect 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 position accordingly. 

It takes about 15 minutes from a "cold" start to make all the adjustments described here after a little experience gained. It is a dramatic experience to be able to move the single-sideband because of a lengthy description of the adjustment procedure. It is 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 pleasant "phone communion" yet developed.
NOTE'S 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-type modulator, it became desirable to have the designer, W1KK, explain these units in further detail for the benefit of the technically inclined readers of Radio 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 complexity 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 component used are not stable enough to hold their values after adjustment. This is especially true for capacitors since variation within 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 AE must be a 200 ± 50 volts (1.55 times the voltage supplied to A). Incidently, the voltage output of each section is equal to the voltage at AE from zero frequency to a matter of megacycles. The design center frequency for the two networks (say, there are actually two) is 400 CPR. The differential phase-shift versus frequency curve is symmetrical about this point and falls to within 1.2 degrees CPR at 225 CPR to 275 CPR, as indicated in Fig. 12. A slight error in setting the reference frequency (390 CPR) 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, D1 and D2 with associated circuits. First, suppose a high frequency signal of a few volts is applied at point B. On the positive crest of signal current, passes through D1, into the center tapped resonant circuit and back to point B in the same direction. Point C naturally tends to go negative while D2 tries to drive the center point of the resonant circuit. No such point, however, because it is the negative direction. But at this time point C will, be at a positive potential because of the “inertia” of the resonant circuit. The net result of the battle between G2 and G1 to cause current to flow in the resonant circuit is a d.c. voltage. No net voltage appears across this circuit at the source frequency and energy is dissipated in the balancing resistor and in G1 and G2. Thus far, we have currents in the resonant circuit, but none at operating frequency. This seems like a long way to go to get nothing, but wait.

Now, let's imagine a bias applied at U. If the voltage at B is positive, G1 will pass more current into the resonant circuit, and G2 will pass less current. This, in effect, unbalances the circuit and a radio frequency voltage will appear across the resonant circuit, with point B in phase with the voltage at R. If the bias voltage at U is negative, G2 passes more current than G1, 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 range) and the radio frequency output voltage will flow in the resonant circuit. As U is varied from 0 to 6 volts, a space of power output is obtained from the resonant circuit in such a manner as to reinforce one set of waveforms and to cancel another. This produces a single-sided suppressed carrier output signal. In the case of CW operation it is a carrier, in the case of SSB Jr.

The function of the balancing resistors (R1 and R2) is to equalize the quadrature, minor differences in the characteristics of the diodes and to balance out stray coupling. Thus, any one balanced modulator is not necessarily perfectly balanced, but the action of two such modulators fed with phase 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 designed for the 75 meter band.
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 every-thing goes as we 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.

**H H H**

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* by W. E. Norgard, W1XJL, 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 com-munication 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 picking 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 in-ver ted 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 any-thing. For example, if you say "General Electric Company" into an inverted speech system, what comes out sounds like "Qwemri Dsmnoeitqen Krim-kins." Conversely, if you say the latter phrase into a normal 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, "meta pro see" says "low path." But you can go even further, as Don and I did. We decided that we could 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:

*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:

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Nonsense had Osputa yamung,
Ut ushing yit yit us seen.
And I view hair hop nasowo yump.
No yang yit six per bay.
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*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!

**H H H**

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, W5JMP, a contributor in the September-October, 1959 *Ham News* (page 13) regarding radio interference from fluorescent lamps, was intrigued by the length, and referred to a homemade filter which might be made consisting of three 0.07 mil 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 mil delta-connected con-densers. One such concern is Barrow Electric Com-pany, and their interference filter has the number 32-37. 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 survey myself with more G-E catalogues. The Q-S unit, by the way, carries the number 28F124.

**H H H**

The reason this lack of the G-E *Ham News* feels shyer or braver is that it contains twelve pages. This is not given to be the standard size of the *Ham News* Jr. rig, I am led to doubt it will give as much in-formation as possible, both in operating and wavelengths. It is possible that one or two issues a year may be twelve pages long, if the material warrants it.

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*Lighthouse* *Leroy*
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 Bell, MT 8
Base: 6H-21 Small Water Getal 6-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 µF
Input ........................................... 13 µF
Output ......................................... 7.5 µF
Grid to Bore (Appendix) ................. 5.8 µF
Grid to Cathode (Appendix) .............. 3.2 µF
Heater to Cathode (Appendix) ........... 10.7 µF

PHYSICAL DIMENSIONS

A B C D

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, 1 wrote them), but 1, 2, 3 and 4 will work. That’s the job. I didn’t peek ahead in the instructions, either.

If you get the tenth of an inch out of building and operating SSB, I, as I did in designing, building and using it, you are in for the most enjoyment you have ever had in ham radio.—W1KUJ