CHAPTER SEVENTEEN

Sideband Transmission

While *single-sideband transmission* (SSB) has attracted significant interest on amateur frequencies only in the past few years, the principles have been recognized and put to use in various commercial applications for many years. Expansion of single-sideband for both commercial and amateur communication has awaited the development of economical components possessing the required characteristics (such as sharp cutoff filters and high stability crystals) demanded by SSB techniques. The availability of such components and precision test equipment now makes possible the economical testing, adjustment and use of SSB equipment on a wider scale than before. Many of the seemingly insurmountable obstacles of past years no longer prevent the amateur from achieving the advantages of SSB for his class of operation.

17-1 Commercial Applications of SSB

Before discussion of amateur SSB equipment, it is helpful to review some of the commercial applications of SSB in an effort to avoid problems that are already solved.

The first and only large scale use of SSB has been for multiplexing additional voice circuits on long distance telephone toll wires. Carrier systems came into wide use during the 30's, accompanied by the development of high Q toroids and copper oxide ring modulators of controlled characteristics.

The problem solved by the carrier system was that of translating the 300-3000 cycle voice band of frequencies to a higher frequency (for example, 40.3 to 43.0 kc.) for transmission on the toll wires, and then to reverse the translation process at the receiving terminal. It was possible in some short-haul equipment to amplitude modulate a 40 kilocycle carrier with the voice frequencies, in which case the resulting signal would occupy a band of frequencies between 37 and 43 kilocycles. Since the transmission properties of wires and cable deteriorate rapidly with increasing frequency, most systems required the bandwidth conservation characteristics of single-sideband transmission. In addition, the carrier wave was generally suppressed to reduce the power handling capability of the repeater amplifiers and diode modulators. A substantial body of literature on the components and circuit techniques of SSB has been generated by the large and continuing development effort to produce economical carrier telephone systems.

The use of SSB for overseas radiotelephony has been practiced for several years though the number of such circuits has been numerically small. However, the economic value of such circuits has been great enough to warrant elaborate station equipment. It is from these stations that the impression has been obtained that SSB is too complicated for all but a corps of engineers and technicians to handle. Components such as lattice filters with 40 or more crystals have suggested astronomical expense.

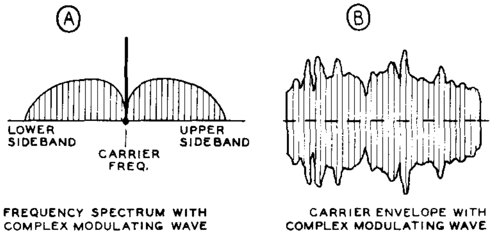


Figure 1
REPRESENTATION OF A CONVENTIONAL AM SIGNAL

More recently, SSB techniques have been used to multiplex large numbers of voice channels on a microwave radio band using equipment principally developed for telephone carrier applications. It should be noted that all production equipment employed in these services uses the *filter method* of generating the single-sideband signal, though there is a wide variation in the types of filters actually used. The SSB signal is generated at a low frequency and at a low level, and then translated and linearly amplified to a high level at the operating frequency.

Considerable development effort has been expended on high level phasing type transmitters wherein the problems of linear amplification are exchanged for the problems of accurately controlled phase shifts. Such equipment has featured automatic tuning circuits, servo-driven to facilitate frequency changing, but no transmitter of this type has been sufficiently attractive to warrant appreciable production.

17-2 Derivation of Single-Sideband Signals

The single-sideband method of communication is, essentially, a procedure for obtaining more efficient use of available frequency spectrum and of available transmitter capability. As a starting point for the discussion of single-sideband signals, let us take a conventional AM signal, such as shown in figure 1, as representing the most common method for transmitting complex intelligence such as voice or music.

It will be noted in figure 1 that there are three distinct portions to the signal: the carrier, and the upper and the lower sideband group. These three portions always are present in a conventional AM signal. Of all these portions the carrier is the least necessary and the most expensive to transmit. It is an actual fact, and it can be proved mathematically (and physically with a highly selective receiver) that the carrier of an AM signal remains unchanged in amplitude, whether it is being modulated or not. Of course the carrier *appears* to be modulated when we observe the modulated signal on a receiving system or indicator which passes a sufficiently wide band that the carrier and the modulation sidebands are viewed at the same time. This apparent change in the amplitude of the carrier with modulation is simply the result of the sidebands beating with the carrier. However, if we receive the signal on a highly selective receiver, and if we modulate the carrier with a sine wave of 3000 to 5000 cycles, we will readily see that the carrier, or either of the sidebands can be tuned in separately; the carrier amplitude, as observed on a signal strength meter, will remain constant, while the amplitude of the sidebands will vary in direct proportion to the modulation percentage.

Elimination of the Carrier and One Sideband It is obvious from the previous discussion that the carrier is superfluous so far as the transmission of intelligence is concerned. It is obviously a convenience, however, since it provides a signal at the receiving end for the sidebands to beat with and thus to reproduce the original modulating signal. It is equally true that the transmission of both sidebands under ordinary conditions is superfluous since identically the same intelligence is contained in both sidebands. Several systems for carrier and sideband elimination will be discussed in this chapter.

Power Advantage of SSB over AM Single sideband is a very efficient form of voice communication by radio. The amount of radio frequency spectrum occupied can be no greater than the frequency range of the audio or speech signal transmitted, whereas other forms of radio transmission require from two to several times as much spectrum space. The r-f power in the transmitted SSB signal is directly proportional to the power in the original audio signal and no strong carrier is transmitted. Except for a weak pilot carrier present in some commercial usage, there is no r-f output when there is no audio input.

The power output rating of a SSB transmitter is given in terms of *peak envelope power* (PEP). This may be defined as the r-m-s power at the crest of the modulation envelope. The peak envelope power of a conventional amplitude modulated signal at 100% modulation is four times the carrier power. The average power input to a SSB transmitter is therefore a very small fraction of the power input to a conventional amplitude modulated transmitter of the same power rating.

Single sideband is well suited for long-range communications because of its spectrum and power economy and because it is less susceptible to the effects of selective fading and interference than amplitude modulation. The principal advantages of SSB arise from the elimination of the high-energy carrier and from further reduction in sideband power permitted by the improved performance of SSB under unfavorable propagation conditions.

In the presence of narrow band man-made interference, the narrower bandwidth of SSB reduces the probability of destructive interference .A statistical study of the distribution of signals on the air versus the signal strength shows that the probability of successful communication will be the same if the SSB power is equal to one-half the power of one of the two a-m sidebands. Thus SSB can give from 0 to 9 db improvement under various conditions when the *total* sideband power is equal in SSB and a-m. In general, it may be assumed that 3 db of the possible 9 db advantage will be realized on the average contact. In this case, the SSB power required for equivalent performance is equal to the power in one of the a-m sidebands. For example, this would rate a 100-watt SSB and a 400 watt (carrier) a-m transmitter as having equal performance. It should be noted that in this comparison it is assumed that the receiver bandwidth is just sufficient to accept the transmitted intelligence in each case.

To help evaluate other methods of comparison the following points should be considered. In conventional amplitude modulation two sidebands are transmitted, each having a peak envelope power equal to ¼-carrier power. For example, a 100-watt a-m signal will have 25-watt peak envelope power in each sideband, or a total of 50 watts. When the receiver detects this signal, the voltages of the two sidebands are added in the detector. Thus the detector output voltage is equivalent to that of a 100-watt SSB signal. This method of comparison says that a 100 watt SSB transmitter is just equivalent to a 100-watt a-m transmitter. This assumption is valid only when the receiver bandwidth used for SSB is the same as that required for amplitude modulation

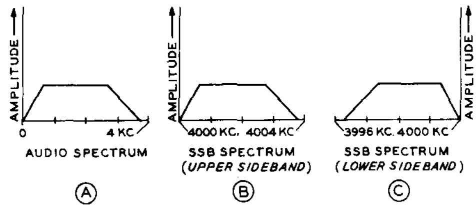


Figure 2
RELATIONSHIP OF AUDIO AND SSB SPECTRUMS
The single sideband components are the same as the original audio components except that the frequency of each is raised by the frequency of the carrier. The relative amplitude of the various components remains the same.

(e.g., 6 kilocycles), when there is no noise or interference other than broadband noise, and if the a-m signal is not degraded by propagation. By using half the bandwidth for SSB reception (e.g., 3 kilocycles) the noise is reduced 3 db so the 100 watt SSB signal becomes equivalent to a 200 watt carrier a-m signal. It is also possible for the a-m signal to be degraded another 3 db on the average due to narrow band interference and poor propagation conditions, giving a possible 4 to 1 power advantage to the SSB signal.

It should be noted that 3 db signal-to-noise ratio is lost when receiving only one sideband of an a-m signal. The narrower receiving bandwidth reduces the noise by 3 db but the 6 db advantage of coherent detection is lost, leaving a net loss of 3 db. Poor propagation will degrade this "one sideband" reception of an a-m signal less than double sideband reception, however. Also under severe narrow band interference conditions (e.g., an adjacent strong signal) the ability to reject all interference on one side of the carrier is a great advantage.

The Nature of a SSB Signal The nature of a single sideband signal is easily visualized by noting that the SSB signal components are exactly the same as the original audio components except that the frequency of each is *raised* by the frequency of the carrier. The relative amplitude of the various components remains the same, however. (The first statement is only true for the upper sideband since the lower sideband frequency components are the *difference* between the carrier and the original audio signal). Figure 2A, B, and C shows how the audio spectrum is simply moved up into the radio spectrum to give the upper sideband. The lower sideband is the same except inverted, as shown in figure 2C. Either sideband may be used. It is apparent that the carrier frequency

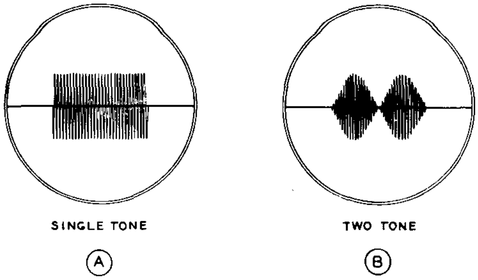


Figure 3
A SINGLE SINE WAVE TONE INPUT TO A SSB TRANSMITTER RESULTS IN A STEADY SINGLE SINE WAVE R-F OUTFIT (A). TWO AUDIO TONES OF EQUAL AMPLITUDE BEAT TOGETHER TO PRODUCE HALF-SINE WAVES AS SHOWN IN (B).

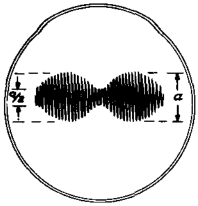


Figure 5
TWO-TONE SSB ENVELOPE WHEN ONE TONE HAS TWICE THE AMPLITUDE OF THE OTHER.

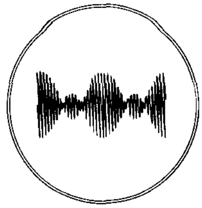


Figure 6
THREE-TONE SSB ENVELOPE WHEN EQUAL TONES OF EQUAL FREQUENCY SPACINGS ARE USED.

of a SSB signal can only be changed by adding or subtracting to the original carrier frequency. This is done by heterodyning, using converter or mixer circuits similar to those employed in a superheterodyne receiver.

It is noted that a single sine wave tone input to a SSB transmitter results in a single steady sine wave r-f ouput, as shown in figure 3A. Since it is difficult to measure the performance of a linear amplifier with a single tone, it has become standard practice to use two tones of equal amplitude for test purposes. The two radio frequencies thus produced beat together to give the SSB envelope shown in figure 3B. This figure has the shape of half sine waves, and from one null to the next represents one full cycle of the difference frequency. How this envelope is generated is shown more fully in figures 4A and 4B. f_1 and f_2 represent the two tone signals. When a vector representing the lower frequency tone signal is used as a reference, the other vector rotates around it as shown, and this action generates the SSB envelope When the two vectors are exactly opposite in phase, the output is zero and this causes the null in the envelope. If one tone has twice the amplitude of the other, the envelope shape is shown in figure 5. Figure 6 shows the SSB envelope of three equal tones of equal frequency spacings and at one particular phase relationship. Figure 7A shows the SSB envelope of four equal tones with equal frequency spacings and at one particular phase relationship. The phase relationships chosen are such that at some instant the vectors representing the several tones are all in phase. Figure 7B shows a SSB envelope of a square wave. *A pure square wave requires infinite bandwidth, so its SSB envelope requires infinite amplitude. This emphasizes the point that the SSB envelope shape is not the same as the original audio wave shape, and usually bears no similarity to it.* This is because the percentage difference between the radio frequencies is very small, even though one audio tone may be several times the other in terms of frequency. Speech clipping as used

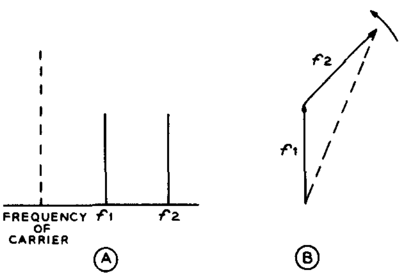


Figure 4
VECTOR REPRESENTATION OF TWO-TONE SSB ENVELOPE

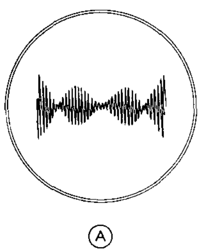


Figure 7A
FOUR TONE SSB ENVELOPE
when equal tones with equal frequency spacings are used

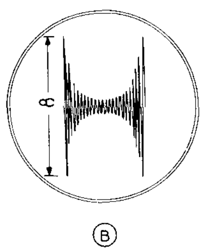


Figure 7B
SSB ENVELOPE OF A SQUARE WAVE.
Peak of wave reaches infinite amplitude.

in amplitude modulation is of no practical value in SSB because the SSB r-f envelopes are so different than the audio envelopes. A heavily clipped wave approaches a square wave and a square wave gives a SSB envelope with peaks of infinite amplitude as shown in figure 7B.

Carrier Frequency Stability Requirements Reception of a SSB signal is accomplished by simply heterodyning the carrier down to zero frequency. (The conversion frequency used in the last heterodyne step is often called the *reinserted carrier*). If the SSB signal is not heterodyned down to exactly zero frequency, each frequency component of the detected audio signal will be high or low by the amount of this error. An error of 10 to 20 c.p s. for speech signals is acceptable from an intelligibility standpoint, but an error of the order of 50 c.p.s. seriously degrades the intelligibility. An error of 20 c.p.s. is not acceptable for the transmission of music, however, because the harmonic relationship of the notes would be destroyed. For example, the harmonics of 220 c.p.s. are 440, 660, 880, etc., but a 10 c.p s. error gives 230, 450, 670, 890, etc., or 210, 430, 650, 870, etc., if the original error is on the other side. This error would destroy the original sound of the tones, and the harmony between the tones.

Suppression of the carrier is common in amateur SSB work, so the combined frequency stabilities of all oscillators in both the transmitting and receiving equipment add together to give the frequency error found in detection. In order to overcome much of the frequency stability problem, it is common commercial practice to transmit a pilot carrier at a reduced amplitude. This is usually 20 db below one tone of a two-tone signal, or 26 db below the peak envelope power rating of the transmitter. This pilot carrier is filtered out from the other signals at the receiver and either amplified and used for the reinserted carrier or used to control the frequency of a local oscillator. By this means, the frequency drift of the carrier is eliminated as an error in detection.

Advantage of SSB with Selective Fading On long distance communication circuits using a-m, selective fading often causes severe distortion and at times makes the signal unintelligible. When one sideband is weaker than the other, distortion results; but when the carrier becomes weak and the sidebands are strong, the distortion is extremely severe and the signal may sound like "monkey chatter." This is because a carrier of at least twice the amplitude of either sideband is necessary to demodulate the signal properly. This can be overcome by using exalted carrier reception in which the carrier is amplified separately and then reinserted before the signal is demodulated or detected. This is a great help, but the reinserted carrier must be very close to the same phase as the original carrier. For example, if the reinserted carrier were 90 degrees from the original source, the a-m signal would be converted to phase modulation and the usual a-m detector would deliver no output.

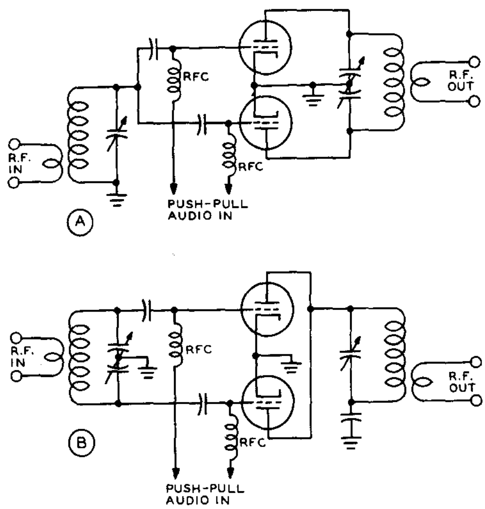


Figure 8

SHOWING TWO COMMON TYPES OF BALANCED MODULATORS

Notice that a balanced modulator changes the circuit condition from single ended to push-pull, or vice versa. Choice of circuit depends upon external circuit conditions since both the (A) and (B) arrangements can give satisfactory generation of a double-sideband suppressed-carrier signal.

The phase of the reinserted carrier is of no importance in SSB reception and by using a strong reinserted carrier, exalted carrier reception is in effect realized. Selective fading with one sideband simply changes the amplitude and the frequency response of the system and very seldom causes the signal to become unintelligible. Thus the receiving techniques used with SSB are those which inherently greatly minimize distortion due to selective fading.

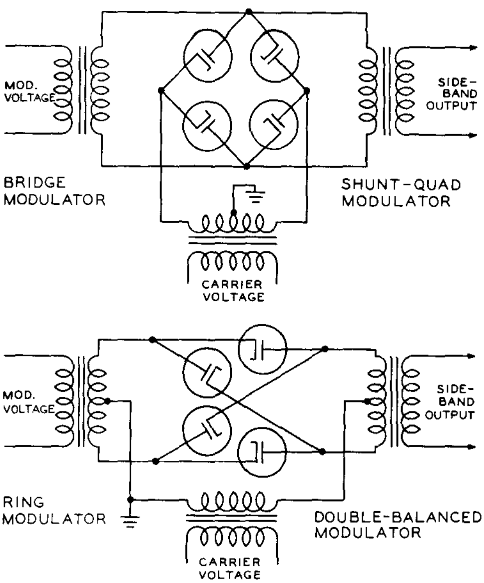


Figure 9
TWO TYPES OF DIODE BALANCED MODULATOR

Such balanced modulator circuits are commonly used in carrier telephone work and in single-sideband systems where the carrier frequency and modulating frequency are relatively close together. Vacuum diodes, copper-oxide rectifiers, or crystal diodes may be used in the circuits.

17-3 Carrier Elimination Circuits

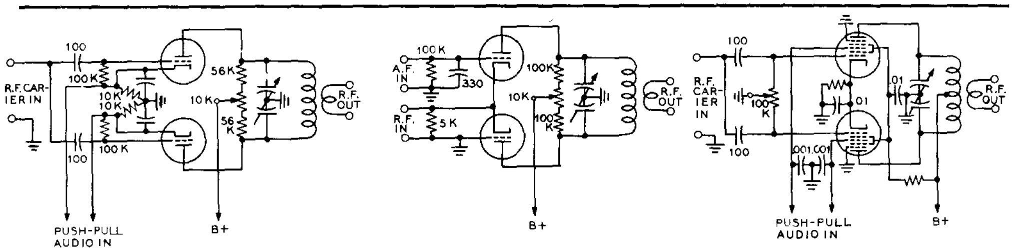
Various circuits may be employed to eliminate the carrier to provide a double sideband signal. A selective filter may follow the carrier elimination circuit to produce a single sideband signal.

Two modulated amplifiers may be connected with the carrier inputs 180° out of phase, and with the carrier outputs in parallel. The carrier will be balanced out of the output circuit, leaving only the two sidebands. Such a circuit is called a *balanced modulator.*

Any non-linear element will produce modulation. That is, if two signals are put in, sum and difference frequencies as well as the original frequencies appear in the output. This phenomenon is objectionable in amplifiers and desirable in modulators or mixers.

In addition to the sum and difference frequencies, other outputs (such as twice one frequency plus the other) may appear. All combinations of all harmonics of each input frequency may appear, but in general these are of decreasing amplitude with increasing order of harmonic. These outputs are usually rejected by selective circuits following the modulator. All modulators are not alike in the magnitude of these higher order outputs. Balanced diode rings operating in the square law region are fairly good and pentagrid converters much poorer. Excessive carrier level in tube mixers will increase the relative magnitude of the higher order outputs. Two types of triode balanced modulators are shown in figure 8, and two types of diode modulators in figure 9. Balanced modulators employing vacuum tubes may be made to work very easily to a point. Circuits may be devised wherein both input signals may be applied to a high impedance grid, simplifying isolation and loading problems. The most important difficulties with these vacuum tube modulator circuits are: (1) Balance is not independent of signal level. (2) Balance drifts with time and environment. (3) The carrier level for low "high-order output" is critical, and (4) Such circuits have limited dynamic range.

A number of typical circuits are shown in figure 10. Of the group the most satisfactory performance is to be had from plate modulated triodes.



Ⓐ PLATE MODULATED BALANCED TRIODE MODULATOR

Ⓑ BALANCED TRIODE MODULATOR WITH SINGLE ENDED INPUT CIRCUITS

Ⓒ BALANCED PENTAGRID CONVERTER MODULATOR

Figure 10
BALANCED MODULATORS

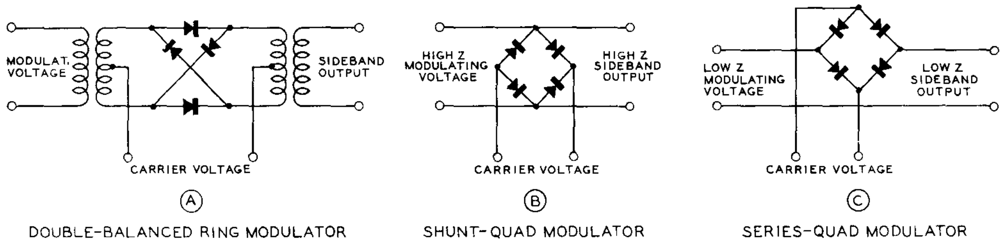


Figure 11
DIODE RING MODULATORS

Diode Ring Modulators Modulation in telephone carrier equipment has been very successfully accomplished with copper-oxide double balanced ring modulators. More recently, germanium diodes have been applied to similar circuits. The basic diode ring circuits are shown in figure 11. The most widely applied is the double balanced ring (A). Both carrier and input are balanced with respect to the output, which is advantageous when the output frequency is not sufficiently different from the inputs to allow ready separation by filters. It should be noted that the carrier must pass through the balanced input and output transformers. Care must be taken in adapting this circuit to minimize the carrier power that will be lost in these elements. The shunt and series quad circuits are usable when the output frequencies are entirely different (i.e.: audio and r.f.). The shunt quad (B) is used with high source and load impedances and the series quad (C) with low source and load impedances. These two circuits may be adapted to use only two diodes, substituting a balanced transformer for one side of the bridge, as shown in figure 12. It should be noted that these circuits present a half-wave load to the carrier source. In applying any of these circuits, r-f chokes and capacitors must be employed to control the path of signal and carrier currents. In the shunt pair, for example, a blocking capacitor is used to prevent the r-f load from shorting the audio input.

To a first approximation, the source and load impedances should be an arithmetical mean of the forward and back resistances of the diodes employed. A workable rule of thumb is that the source and load impedances be ten to twenty times the forward resistance for semi-conductor rings. The high frequency limit of operation in the case of junction and copper-oxide diodes may be appreciably extended by the use of very low source and load impedances.

Copper-oxide diodes suitable for carrier work are normally manufactured to order. They offer no particular advantage to the amateur, though their excellent long-term stability is important in commercial applications. Rectifier types intended to be used as meter rectifiers are not likely to have the balance or high frequency response desirable in amateur SSB transmitters.

Vacuum diodes such as the 6AL5 may be used as modulators. Balancing the heater-cathode capacity is a major difficulty except when the 6AL5 is used at low source and load impedance levels. In addition, contact potentials of the order of a few tenths of a volt may also disturb low level applications (figure 13).

The double diode circuits appear attractive, but in general it is more difficult to balance a transformer at carrier frequency than an additional pair of diodes. Balancing potentiometers may be employed, but the actual cause of the unbalance is far more subtile, and cannot be adequately corrected with a single adjustment.

A signal produced by any of the above circuits may be classified as a *double sideband, suppressed-carrier* signal.

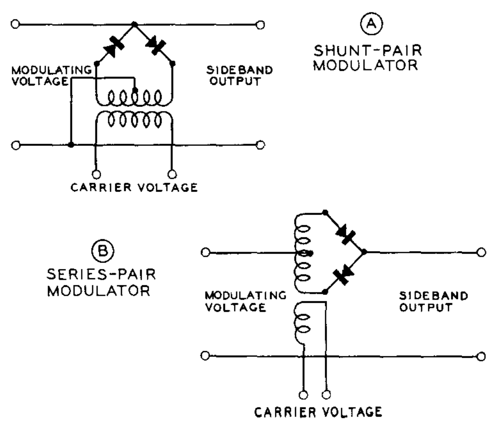
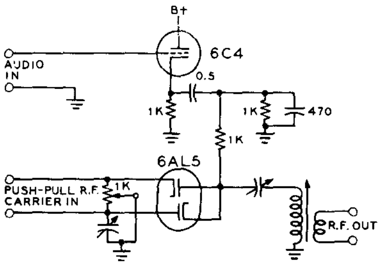
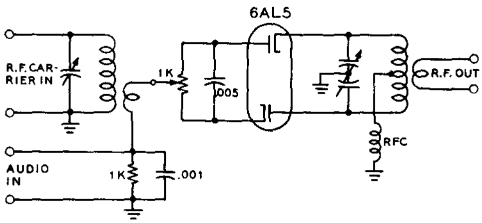


Figure 12
DOUBLE-DIODE PAIRED MODULATORS



Ⓐ SERIES-BALANCED DIODE MODULATOR USING 6AL5 TUBE



Ⓑ RING-DIODE MODULATOR USING 6AL5 TUBE

Figure 13
VACUUM DIODE MODULATOR CIRCUITS

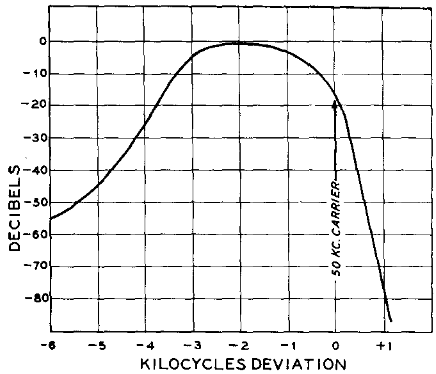


Figure 15
BANDPASS CHARACTERISTIC OF BURNELL S-15000 SINGLE SIDEBAND FILTER

17-4 Generation of Single-Sideband Signals

In general, there are two commonly used methods by which a single-sideband signal may be generated. These systems are: (1) The Filter Method, and (2) The Phasing Method. The systems may be used singly or in combination, and either method, in theory, may be used at the operating frequency of the transmitter or at some other frequency with the signal at the operating frequency being obtained through the use of frequency changers (mixers).

The Filter Method The filter method for obtaining a SSB signal is the classic method which has been in use by the telephone companies for many years both for land-line and radio communications. The mode of operation of the filter method is diagrammed in figure 14, in terms of components and filters which normally would be available to the amateur or experimenter. The output of the speech amplifier passes through a conventional speech filter to limit the frequency range of the speech to about 200 to 3000 cycles. This signal then is fed to a balanced modulator along with a 50,000-cycle first carrier from a self-excited oscillator. A low-frequency balanced modulator of this type most conveniently may be made up of four diodes of the vacuum or crystal type cross connected in a balanced bridge or ring modulator circuit. Such a modulator passes only the sideband components resulting from the sum and difference between the two signals being fed to the balanced modulator. The audio signal and the 50-kc. carrier signal from the oscillator both cancel out in the balanced modulator so that a band of frequencies between 47 and 50 kc. and another band of frequencies between 50 and 53 kc. appear in the output.

The signals from the first balanced modulator are then fed through the most critical

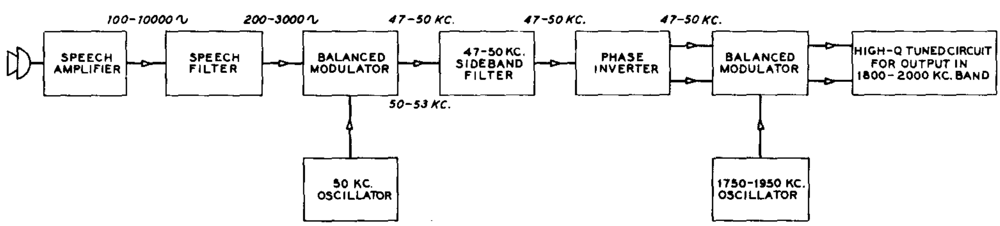


Figure 14
BLOCK DIAGRAM OF FILTER EXCITER EMPLOYING A 50-K.C. SIDEBAND FILTER

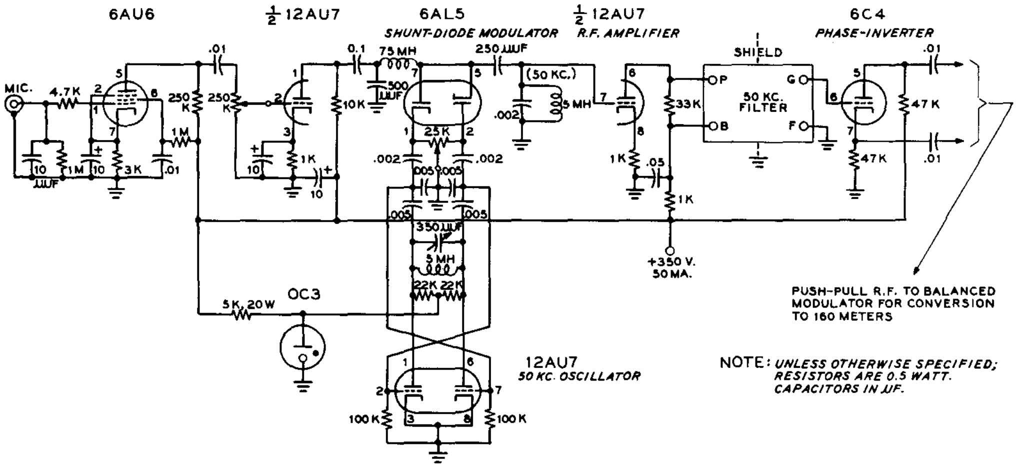


Figure 16
OPERATIONAL CIRCUIT FOR SSB EXCITER USING THE BURNELL 50-KC. SIDEBAND FILTER

component in the whole system—the first sideband filter. It is the function of this first sideband filter to separate the desired 47 to 50 kc. sideband from the unneeded and undesired 50 to 53 kc. sideband. Hence this filter must have low attenuation in the region between 47 and 50 kc., a very rapid slope in the vicinity of 50 kc., and a very high attenuation to the sideband components falling between 50 and 53 kilocycles.

Burnell & Co., Inc., of Yonkers, New York produce such a filter, designated as *Burnell S-15,000.* The passband of this filter is shown in figure 15.

Appearing, then, at the output of the filter is a single sideband of 47 kc. to 50 kc. This sideband may be passed through a phase inverter to obtain a balanced output, and then fed to a balanced mixer. A local oscillator operating in the range of 1750 kc. to 1950 kc. is used as the conversion oscillator. Additional conversion stages may now be added to translate the SSB signal to the desired frequency. Since only linear amplification may be used, it is not possible to use frequency multiplying stages. Any frequency changing must be done by the beating-oscillator technique. An operational circuit of this type of SSB exciter is shown in figure 16.

A second type of filter-exciter for SSB may be built around the *Collins* Mechanical Filter. Such an exciter is diagrammed in figure 17. Voice frequencies in the range of 200-3000 cycles are amplified and fed to a low impedance phase-inverter to furnish balanced audio. This audio, together with a suitably chosen r-f signal, is mixed in a ring modulator, made up of small germanium diodes. Depending upon the choice of frequency of the r-f oscillator, either the upper or lower sideband may be applied to the input of the mechanical filter. The carrier, to some extent, has been rejected by the ring modulator. Additional carrier rejection is afforded by the excellent passband

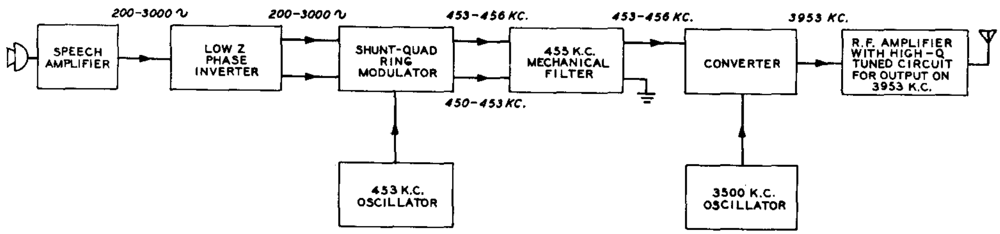


Figure 17
BLOCK DIAGRAM OF FILTER EXCITER EMPLOYING A 455-KC. MECHANICAL FILTER FOR SIDEBAND SELECTION

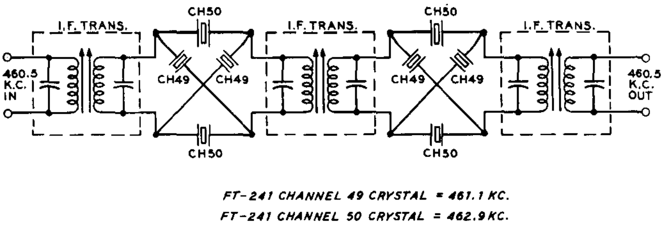
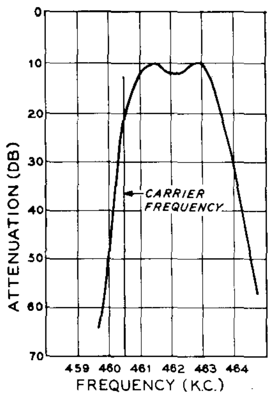


Figure 18
SIMPLE CRYSTAL LATTICE FILTER



characteristics of the mechanical filter. For simplicity, the mixing and filtering operation usually takes place at a frequency of 455 kilocycles. The single-sideband signal appearing at the output of the mechanical filter may be translated directly to a higher operating frequency. Suitable tuned circuits must follow the conversion stage to eliminate the signal from the conversion oscillator.

Wave Filters The heart of a filter-type SSB exciter is the sideband filter. Conventional coils and capacitors may be used to construct a filter based upon standard wave filter techniques. The Q of the filter inductances must be high when compared with the reciprocal of the fractional bandwidth. If a bandwidth of 3 kc. is needed at a carrier frequency of 50 kc., the bandwidth expressed in terms of the carrier frequency is 3/50 or 6%. This is expressed in terms of fractional bandwidth as 1/16. For satisfactory operation, the

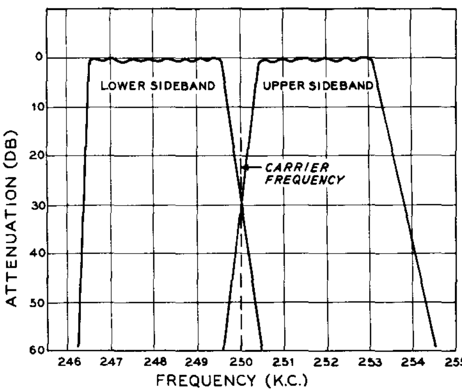


Figure 19
PASSBAND OF LOWER AND UPPER SIDEBAND MECHANICAL FILTER

Q of the filter inductances should be 10 times the reciprocal of this, or 160. Appropriate Q is generally obtained from toroidal inductances, though there is some possibility of using iron core solenoids between 10 kc. and 20 kc. A characteristic impedance below 1000 ohms should be selected to prevent distributed capacity of the inductances from spoiling overall performance. Paper capacitors intended for bypass work may not be trusted for stability or low loss and should not be used in filter circuits. Care should be taken that the levels of both accepted and rejected signals are low enough so that saturation of the filter inductances does not occur.

Crystal Filters The best known filter responses have been obtained with crystal filters. Types designed for program carrier service cut-off 80 db in less than 50 cycles. More than 80 crystals are used in this type of filter. The crystals are cut to control reactance and resistance as well as the resonant frequency. The circuits used are based on full lattices.

The war-surplus low frequency crystals may be adapted to this type of filter with some success. Experimental designs usually synthesize a selectivity curve by grouping sharp notches at the side of the passband. Where the width of the passband is greater than twice the spacing of the series and parallel resonance of the crystals, special circuit techniques must be used. A typical crystal filter using these surplus crystals, and its approximate passband is shown in figure 18.

Mechanical Filters Filters using mechanical resonators have been studied by a number of companies and are offered commercially by the *Collins Radio Co.* They are available in a variety of bandwidths

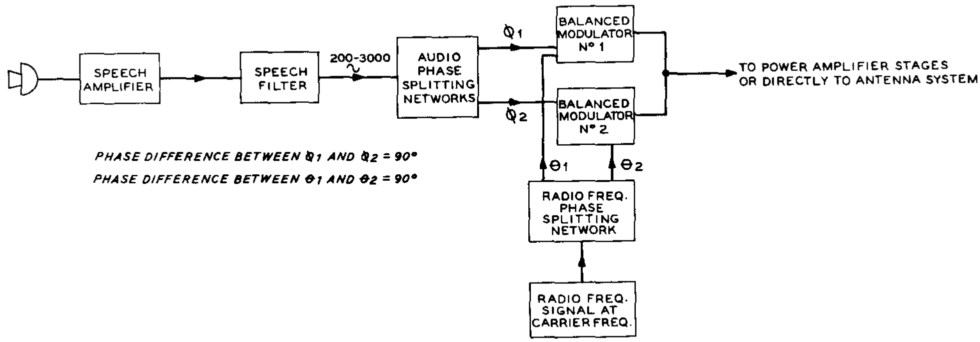


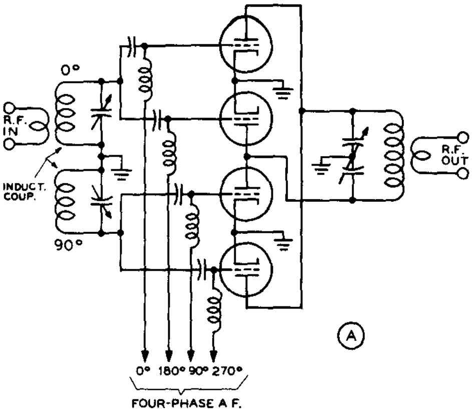
Figure 20
BLOCK DIAGRAM OF THE "PHASING" METHOD
The phasing method of obtaining a single-sideband signal is simpler than the filter system in regard to the number of tubes and circuits required. The system is also less expensive in regard to the components required, but is more critical in regard to adjustments for the transmission of a pure single-sideband signal.

at center frequencies of 250 kc. and 455 kc. The 250 kc. series is specifically intended for sideband selection. The selectivity attained by these filters is intermediate between good LC filters at low center frequencies and engineered quartz crystal filters. A passband of two 250 kc. filters is shown in figure 19. In application of the mechanical filters some special precautions are necessary. The driving and pick-up coils should be carefully resonated to the operating frequency. If circuit capacities are unknown, trimmer capacitors should be used across the coils. Maladjustment of these tuned coils will increase insertion loss and the peak-to-valley ratio. On high impedance filters (ten to twenty thousand ohms) signals greater than 2 volts at the input should be avoided. D-c should be blocked out of the end coils. While the filters are rated for 5 ma. of coil current, they are not rated for d-c plate voltage.

The Phasing System There are a number of points of view from which the operation of the phasing system of SSB generation may be described. We may state that we generate two double-sideband suppressed carrier signals, each in its own balanced modulator, that both the r-f phase and the audio phase of the two signals differ by 90 degrees, and that the outputs of the two balanced modulators are added with the result that one sideband is increased in amplitude and the other one is cancelled. This, of course, is a true description of the action that takes place. But it is much easier to consider the phasing system as a method simply of adding (or of subtracting) the desired modulation frequency and the nominal carrier frequency. The carrier frequency of course is not transmitted, as is the case with all SSB transmissions, but only the sum or the difference of the modulation band from the nominal carrier is transmitted (figure 20).

The phasing system has the obvious advantage that all the electrical circuits which give rise to the single sideband can operate in a practical transmitter at the nominal output frequency of the transmiter. That is to say that if we desire to produce a single sideband whose nominal carrier frequency is 3.9 Mc., the balanced modulators are fed with a 3.9-Mc. signal and with the audio signal from the phase splitters. It is not necessary to go through several frequency conversions in order to obtain a sideband at the desired output frequency, as in the case with the filter method of sideband generation.

Assuming that we feed a speech signal to the balanced modulators along with the 3900-kc. carrier (3.9 Mc.) we will obtain in the output of the balanced modulators a signal which is either the sum of the carrier signal and the speech band, or the difference between the carrier and the speech band. Thus if our speech signal covers the band from 200 to 3000 cycles, we will obtain in the output a band of frequencies from 3900.2 to 3903 kc. (the sum of the two, or the "upper" sideband), or a band from 3897 to 3899.8 kc. (the difference between the two or the "lower" sideband). A further advantage of the phasing system of sideband generation is the fact that it is a very simple matter to select either the upper sideband or the lower sideband for transmission. A simple double-pole double-throw reversing switch in two of the four audio leads to the balanced modulators is all that is required.



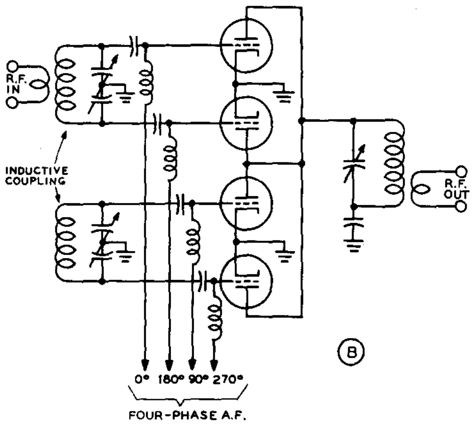


Figure 21
TWO CIRCUITS FOR SINGLE SIDEBAND GENERATION BY THE PHASING METHOD.
The circuit of (A) offers the advantages of simplicity in the single-ended input circuits plus a push-pull output circuit. Circuit (B) requires double-ended input circuits but allows all the plates to be connected in parallel for the output circuit.

High-Level Phasing Vs. Low-Level Phasing The plate-circuit efficiency of the four tubes usually used to make up the two balanced modulators of the phasing system may run as high as 50 to 70 per cent, depending upon the operating angle of plate current flow. Hence it is possible to operate the double balanced modulator directly into the antenna system as the output stage of the transmitter.

The alternative arrangement is to generate the SSB signal at a lower level and then to amplify this signal to the level desired by means of class A or class B r-f power amplifiers. If the SSB signal is generated at a level of a few milliwatts it is most common to make the first stage in the amplifier chain a class A amplifier, then to use one or more class B linear amplifiers to bring the output up to the desired level.

Balanced Modulator Circuits Illustrated in figure 8 are the two basic balanced modulator circuits which give good results with a radio frequency carrier and an audio modulating signal. Note that one push-pull and one single ended tank circuit is required, but that the push-pull circuit may be placed either in the plate or the grid circuit. Also, the audio modulating voltage always is fed into the stage in push-pull, and the tubes normally are operated Class A.

When combining two balanced modulators to make up a double balanced modulator as used in the generation of an SSB signal by the phasing system, only one plate circuit is required for the two balanced modulators. However, separate grid circuits are required since the grid circuits of the two balanced modulators operate at an r-f phase difference of 90 degrees. Shown in figure 21 are the two types of double balanced modulator circuits used for generation of an SSB signal. Note that the circuit of figure 21A is derived from the balanced modulator of figure 8A, and similarly figure 21B is derived from figure 8B.

Another circuit that gives excellent performance and is very easy to adjust is shown in figure 22. The adjustments for carrier balance are made by adjusting the potentiometer for voltage balance and then the small variable capacitor for exact phase balance of the balanced carrier voltage feeding the diode modulator.

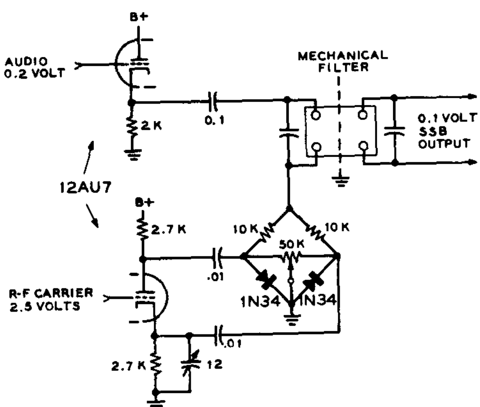


Figure 22
BALANCED MODULATOR FOR USE WITH MECHANICAL FILTER

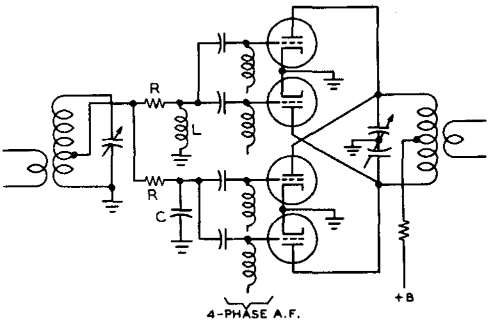


Figure 23
LOW-Q R-F PHASE-SHIFT NETWORK
The r-f phase-shift system illustrated above is convenient in a case where it is desired to make small changes in the operating frequency of the system without the necessity of being precise in the adjustment of two coupled circuits as used for r-f phase shift in the circuit of figure 21.

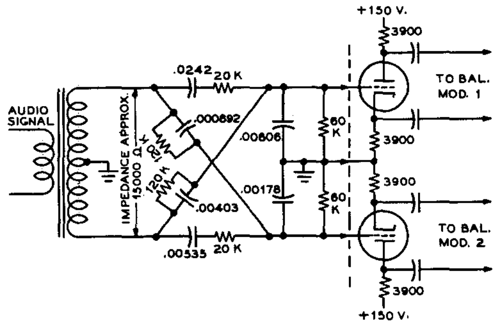


Figure 24
DOME AUDIO-PHASE-SHIFT NETWORK
This circuit arrangement is convenient for obtaining the audio phase shift when it is desired to use a minimum of circuit components and tube elements.

Radio-Frequency Phasing A single-sideband generator of the phasing type requires that the two balanced modulators be fed with r-f signals having a 90-degree phase difference. This r-f phase difference may be obtained through the use of two loosely coupled resonant circuits, such as illustrated in figures 21A and 21B. The r-f signal is coupled directly or inductively to one of the tuned circuits, and the coupling between the two circuits is varied until, at resonance of both circuits, the r-f voltages developed across each circuit have the same amplitude and a 90-degree phase difference.

The 90-degree r-f phase difference also may be obtained through the use of a low-Q phase shifting network, such as illustrated in figure 23; or it may be obtained through the use of a lumped-constant quarter-wave line. The low-Q phase-shifting system has proved quite practicable for use in single-sideband systems, particularly on the lower frequencies. In such an arrangement the two resistances R have the same value, usually in the range between 100 and a few thousand ohms. Capacitor C, in shunt with the input capacitances of the tubes and circuit capacitances, has a reactance at the operating frequency equal to the value of the resistor R. Also, inductor L has a *net* inductive reactance equal in value at the operating frequency to resistance R.

The inductance chosen for use at L must take into account the cancelling effect of the input capacitance of the tubes and the circuit capacitance; hence the inductance should be variable and should have a *lower* value of inductance than that value of inductance which would have the same reactance as resistor R. Inductor L may be considered as being made up of two values of inductance in parallel; (a) a value of inductance which will resonate at the operating frequency with the circuit and tube capacitances, and (b) the value of inductance which is equal in reactance to the resistance R. In a network such as shown in figure 23, equal and opposite 45-degree phase shifts are provided by the RL and RC circuits, thus providing a 90-degree phase difference between the excitation voltages applied to the two balanced modulators.

Audio-Frequency Phasing The audio-frequency phase-shifting networks used in generating a single-sideband signal by the phasing method usually are based on those described by Dome in an article in the December, 1946, *Electronics.* A relatively simple network for accomplishing the 90-degree phase shift over the range from 160 to 3500 cycles is illustrated in figure 24. The values of resistance and capacitance must be carefully checked to insure minimum deviation from a 90-degree phase shift over the 200 to 3000 cycle range.

Another version of the Dome network is shown in figure 25. This network employs three 12AU7 tubes and provides balanced output for the two balanced modulators. As with the previous network, values of the resistances within the network must be held to very close tolerances. It is necessary to restrict the speech range to 300 to 3000 cycles with this network. Audio frequencies outside this range will not have the necessary phase-shift at the output

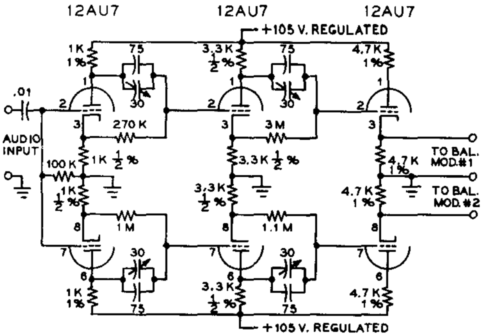


Figure 25
A VERSION OF THE DOME AUDIO-PHASE-SHIFT NETWORK

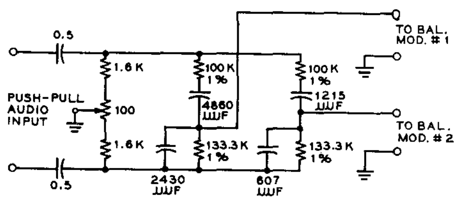


Figure 26
PASSIVE AUDIO-PHASE-SHIFT NETWORK, USEFUL OVER RANGE OF 300 TO 3000 CYCLES.

of the network and will show up as spurious emissions on the sideband signal, and also in the region of the rejected sideband. A low-pass 3500 cycle speech filter, such as the *Chicago Transformer Co. LPF-2* should be used ahead of this phase-shift network.

A passive audio phase-shift network that employs no tubes is shown in figure 26. This network has the same type of operating restrictions as those described above. Additional information concerning phase-shift networks will be found in *Single Sideband Techniques* published by the Cowan Publishing Corp., New York, and *The Single Sideband Digest* published by the American Radio Relay League. A comprehensive sideband review is contained in the December, 1956 issue of *Proceedings of the I.R.E.*

Comparison of Filter and Phasing Methods of SSB Generation Either the filter or the phasing method of single-sideband generation is theoretically capable of a high degree of performance.

In general, it may be said that a high degree of unwanted signal rejection may be attained with less expense and circuit complexity with the filter method. The selective circuits for rejection of unwanted frequencies operate at a relativly low frequency, are designed for this one frequency and have a relatively high order of Q. Carrier rejection of the order of 50 db or so may be obtained with a relatively simple filter and a balanced modulator, and unwanted sideband rejection in the region of 60 db is economically possible.

The phasing method of SSB generation exchanges the problems of high-Q circuits and linear amplification for the problems of accurately controlled phase-shift networks. If the phasing method is employed on the actual transmitting frequency, change of frequency must be accompanied by a corresponding rebalance of the phasing networks. In addition, it is difficult to obtain a phase balance with ordinary equipment within 2% over a band of audio frequencies. This means that carrier suppression is limited to a maximum of 40 db or so. However, when a relatively simple SSB transmitter is needed for spot frequency operation, a phasing unit will perform in a satisfactory manner.

Where a high degree of performance in the SSB exciter is desired, the filter method and the phasing method may be combined. Through the use of the phasing method in the first balanced modulator those undesired sideband components lying within 1000 cycles of the carrier may be given a much higher degree of rejection than is attainable with the filter method alone, with any reasonable amount of complexity in the sideband filter. Then the sideband filter may be used in its normal way to attain very high attenuation of all undesired sideband components lying perhaps further than 500 cycles away from the carrier, and to restrict the sideband width on the *desired* side of the carrier to the specified frequency limit.

17-5 Single Sideband Frequency Conversion Systems

In many instances the band of sideband frequencies generated by a low level SSB transmitter must be heterodyned up to the desired carrier frequency. In receivers the circuits which perform this function are called *converters* or *mixers*. In sideband work they are usually termed *mixers* or *modulators*.

Mixer Stages One circuit which can be used for this purpose employs a receiving-type mixer tube, such as the 6BE6. The output signal from the SSB generator is fed into the #1 grid and the conversion fre-

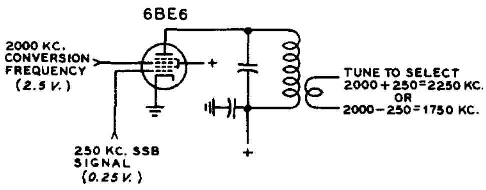


Figure 27
PENTAGRID MIXER CIRCUIT FOR SSB FREQUENCY CONVERSION

quency into the #3 grid. This is the reverse of the usual grid connections, but it offers about 10 db improvement in distortion. The plate circuit is tuned to select the desired output frequency product. Actually, the output of the mixer tube contains all harmonics of the two input signals and all possible combinations of the sum and difference frequencies of all the harmonics. In order to avoid distortion of the SSB signal, it is fed to the mixer at a low level, such as 0.1 to 0.2 volts. The conversion frequency is fed in at a level about 20 db higher, or about 2 volts. By this means, harmonics of the incoming SSB signal generated in the mixer tube will be very low. Usually the desired output frequency is either the sum or the difference of the SSB generator carrier frequency and the conversion frequency. For example, using a SSB generator carrier frequency of 250 kc. and a conversion injection frequency of 2000 kc. as shown in figure 27, the output may be tuned to select either 2250 kc. or 1750 kc.

Not only is it necessary to select the desired mixing product in the mixer output but also the undesired products must be highly attenuated to avoid having spurious output signals from the transmitter. In general, all spurious signals that appear within the assigned frequency channel should be at least 60 db below the desired signal, and those appearing outside of the assigned frequency channel at least 80 db below the signal level.

When mixing 250 kc. with 2000 kc. as in the above example, the desired product is the 2250 kc. signal, but the 2000 kc. injection frequency will appear in the output about 20 db stronger than the desired signal. To reduce it to a level 80 db below the desired signal means that it must be attenuated 100 db.

The principal advantage of using balanced modulator mixer stages is that the injection frequency theoretically does not appear in the output. In practice, when a considerable frequency range must be tuned by the balanced modulator and it is not practical to trim the

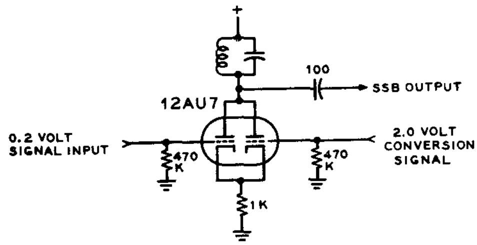


Figure 28
TWIN TRIODE MIXER CIRCUIT FOR SSB FREQUENCY CONVERSION

push-pull circuits and the tubes into exact amplitude and phase balance, about 20 db of injection frequency cancellation is all that can be depended upon. With suitable trimming adjustments the cancellation can be made as high as 40 db, however, in fixed frequency circuits.

The Twin Triode Mixer The mixer circuit shown in figure 28 has about 10 db lower distortion than the conventional 6BE6 converter tube. It has a lower voltage gain of about unity and a lower output impedance which loads the first tuned circuit and reduces its selectivity. In some applications the lower gain is of no consequence but the lower distortion level is important enough to warrant its use in high performance equipment. The signal-to-distortion ratio of this mixer is of the order of 70 db compared to approximately 60 db for a 6BE6 mixer when the level of each of two tone signals is 0.5 volt. With stronger signals, the 6BE6 distortion increases very rapidly, whereas the 12AU7 distortion is much better comparatively.

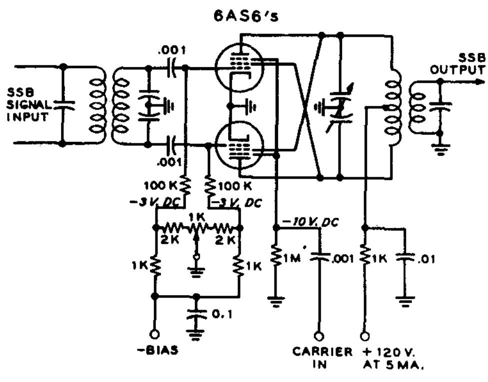


Figure 29
BALANCED MODULATOR CIRCUIT FOR SSB FREQUENCY CONVERSION

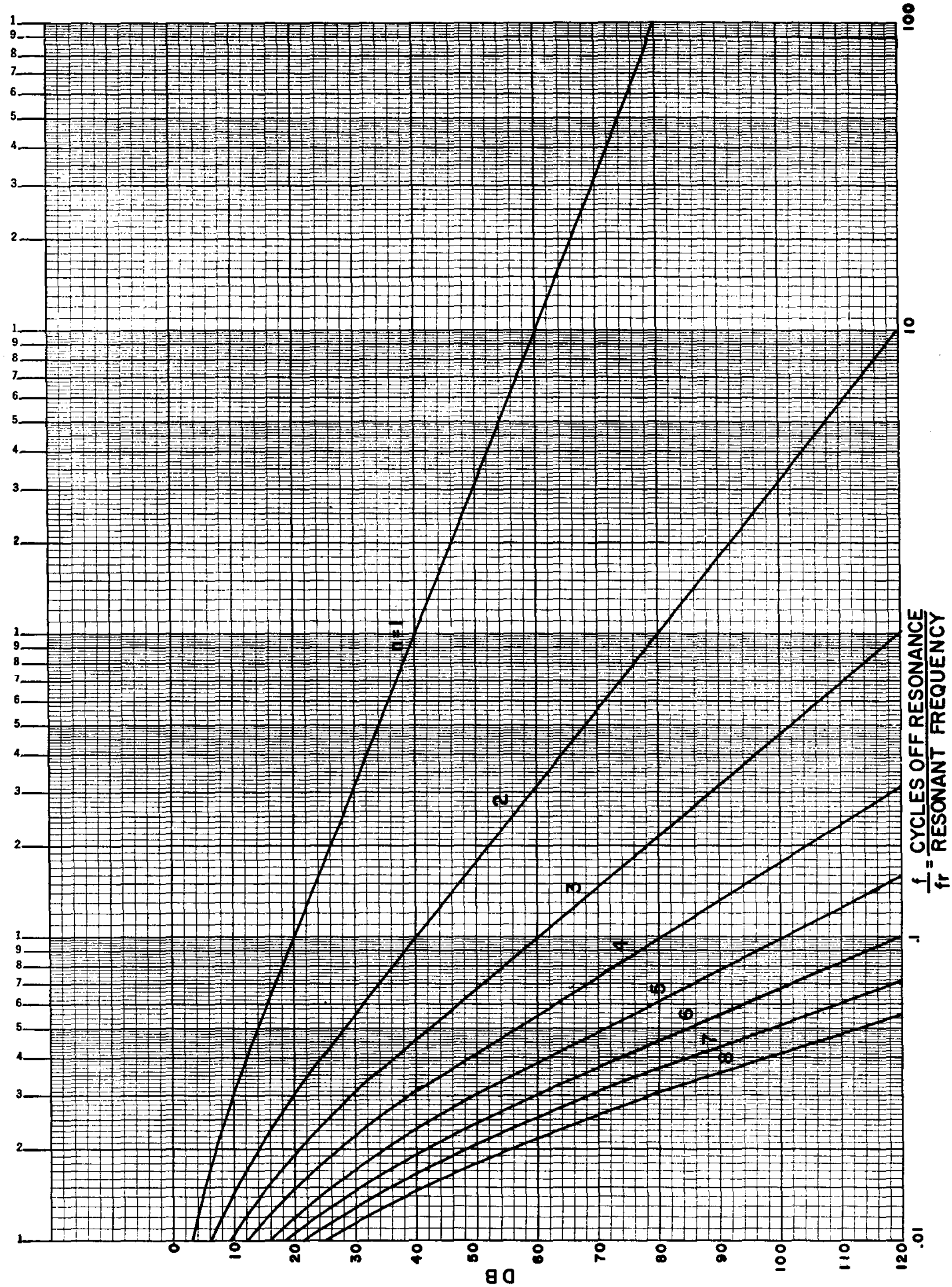


Figure 30
RESPONSE OF "N" NUMBER OF TUNED CIRCUITS, ASSUMING EACH CIRCUIT Q IS 50

In practical equipment where the injection frequency is variable and trimming adjustments and tube selection cannot be used, it may be easier and more economical to obtain this extra 20 db of attenuation by using an extra tuned circuit in the output than by using a balanced modulator circuit. A balanced modulator circuit of interest is shown in figure 29, providing a minimum of 20 db of carrier attenuation with no balancing adjustment.

Selective Tuned Circuits The selectivity requirements of the tuned circuits following a mixer stage often become quite severe. For example, using an input signal at 250 kc. and a conversion injection frequency of 4000 kc. the desired output may be 4250 kc. Passing the 4250 kc. signal and the associated sidebands without attenuation and realizing 100 db of attenuation at 4000 kc. (which is only 250 kc. away) is a practical example. Adding the requirement that this selective circuit must tune from 2250 kc. to 4250 kc. further complicates the basic requirement. The best solution is to cascade a number of tuned circuits. Since a large number of such circuits may be required, the most practical solution is to use permeability tuning, with the circuits tracked together. An example of such circuitry is found in the *Collins KWS-1* sideband transmitter.

If an amplifier tube is placed between each tuned circuit, the overall response will be the sum of one stage multiplied by the number of stages (assuming identical tuned circuits). Figure 30 is a chart which may be used to determine the number of tuned circuits required for a certain degree of attenuation at some nearby frequency. The Q of the circuits is assumed to be 50, which is normally realized in small permeability tuned coils. The number of tuned circuits with a Q of 50 required for providing 100 db of attenuation at 4000 kc. while passing 4250 kc. may be found as follows:

Δf is $4250-4000=250$ kc.

f_r is the resonant frequency, 4250 kc.

$$\text{and } \frac{\Delta f}{f_r} = \frac{250}{4250} = 0.059$$

The point on the chart where .059 intersects 100 db is between the curves for 6 and 7 tuned circuits, so 7 tuned circuits are required.

Another point which must be considered in practice is the tuning and tracking error of the circuits. For example, if the circuits were actually tuned to 4220 kc. instead of 4250 kc., the $\frac{\Delta f}{f_r}$ would be $\frac{220}{4220}$ or 0.0522. Checking the curves shows that 7 circuits would just barely provide 100 db of attenuation. This illustrates the need for very accurate tuning and tracking in circuits having high attenuation properties.

Coupled Tuned Circuits When as many as 7 tuned circuits are required for proper attenuation, it is not necessary to have the gain that 6 isolating amplifier tubes would provide. Several vacuum tubes can be eliminated by using two or three coupled circuits between the amplifiers. With a coefficient of coupling between circuits 0.5 of critical coupling, the overall response is very nearly the same as isolated circuits. The gain through a pair of circuits having 0.5 coupling is only eight-tenths that of two critically coupled circuits, however. If critical coupling is used between two tuned circuits, the nose of the response curve is broadened and about 6 db is lost on the skirts of each pair of critically coupled circuits. In some cases it may be necessary to broaden the nose of the response curve to avoid adversely affecting the frequency response of the desired passband. Another tuned circuit may be required to make up for the loss of attenuation on the skirts of critically coupled circuits.

Frequency Conversion Problems The example in the previous section shows the difficult selectivity problem encountered when strong undesired signals appear near the desired frequency. A high frequency SSB transmitter may be required to operate at any carrier frequency in the range of 1.75 Mc. to 30 Mc. The problem is to find a practical and economical means of heterodyning the generated SSB frequency to any carrier frequency in this range. There are many modulation products in the output of the mixer and a frequency scheme must be found that will not have undesired output of appreciable amplitude at or near the desired signal. When tuning across a frequency range some products may "cross over" the desired frequency. These undesired crossover frequencies should be at least 60 db below the desired signal to meet modern standards. The amplitude of the undesired products depends upon the particular characteristics of the mixer and the particular order of the product. In general, most products of the 7th order and higher will be at least 60 db down. Thus any cross-

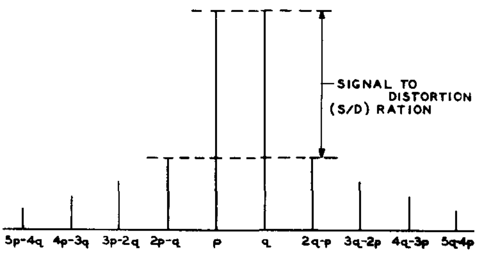


Figure 31
SSB DISTORTION PRODUCTS, SHOWN UP TO NINTH ORDER

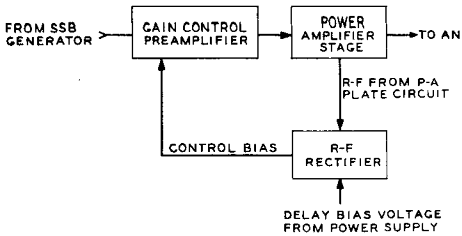


Figure 32
BLOCK DIAGRAM OF AUTOMATIC LOAD CONTROL (A.L.C.) SYSTEM

over frequency lower than the 7th must be avoided since there is no way of attenuating them if they appear within the desired pass-band. The *General Electric Ham News,* volume 11 #6 of Nov.-Dec., 1956 covers the subject of spurious products and incorporates a "mix-selector" chart that is useful in determining spurious products for various different mixing schemes.

In general, for most applications when the intelligence bearing frequency is lower than the conversion frequency, it is desirable that the ratio of the two frequencies be between 5 to 1 and 10 to 1. This is a compromise between avoiding low order harmonics of this signal input appearing in the output, and minimizing the selectivity requirements of the circuits following the mixer stage.

17-6 Distortion Products Due to Nonlinearity of R-F Amplifiers

When the SSB envelope of a *voice* signal is distorted, a great many new frequencies are generated. These represent all of the possible combinations of the sum and difference frequencies of all harmonics of the original frequencies. For purposes of test and analysis, two equal amplitude tones are used as the SSB audio source. Since the SSB radio frequency amplifiers use tank circuits, all distortion products are filtered out except those which lie close to the desired frequencies. These are all odd order products; third order, fifth order, etc.. The third order products are 2p-q and 2q-p where p and q represent the two SSB r-f tone frequencies. The fifth order products are 3p-2q and 3q-2p. These and some higher order products are shown in figure 31. It should be noted that the frequency spacings are always equal to the difference frequency of the two original tones. Thus when a SSB amplifier is badly overloaded, these spurious frequencies can extend far outside the original channel width and cause an unintelligible "splatter" type of interference in adjacent channels. This is usually of far more importance than the distortion of the original tones with regard to intelligibility or fidelity. To avoid interference in another channel, these distortion products should be down at least 40 db below adjacent channel signal. Using a two-tone test, the distortion is given as the ratio of the amplitude of one test tone to the amplitude of a third order product. This is called the *signal-to-distortion ratio* (S/D) and is usually given in decibels. The use of feedback r-f amplifiers make S/D ratios of greater than 40 db possible and practical.

Automatic Load Control Two means may be used to keep the amplitude of these distortion products down to acceptable levels. One is to design the amplifier for excellent linearity over its amplitude or power range. The other is to employ a means of limiting the amplitude of the SSB envelope to the capabilities of the amplifier. *An automatic load control ssytem* (ALC) may be used to accomplish this result. It should be noted that the r-f wave shapes of the SSB signal are always sine waves because the tank circuits make them so. It is the *change in gain* with signal level in an amplifier that distorts the SSB envelope and generates unwanted distortion products. An ALC system may be used to limit the input signal to an amplifier to prevent a change in gain level caused by excessive input level.

The ALC system is adjusted so the power amplifier is operating near its maximum power capability and at the same time is protected from being over-driven. In amplitude modulated systems it is common to use speech compressors and speech clipping systems to perform this function. These methods are not

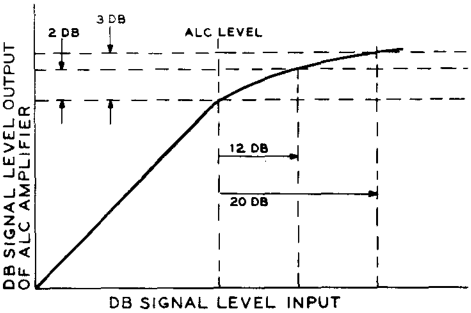


Figure 33
PERFORMANCE CURVE OF A.L.C. CIRCUIT

equally useful in SSB. The reason for this is that the SSB envelope is different from the audio envelope and the SSB peaks do not necessarily correspond with the audio peaks as explained earlier in this chapter. For this reason a "compressor" of some sort located between the SSB generator and the power amplifier is most effective because it is controlled by SSB envelope peaks rather than audio peaks. Such a "SSB signal compressor" and the means of obtaining its control voltage comprises a satisfactory ALC system.

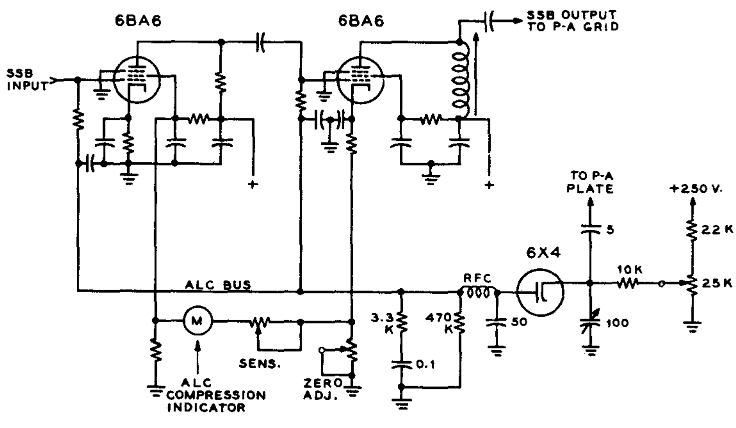
The ALC Circuit A block diagram of an ALC circuit is shown in figure 32. The compressor or gain control part of this circuit uses one or two stages of remote cutoff tubes such as 6BA6, operating very similarly to the intermediate frequency stages of a receiver having automatic volume control.

The grid bias voltage which controls the gain of the tubes is obtained from a voltage detector circuit connected to the power amplifier tube plate circuit. A large delay bias is used so that no gain reduction takes place until the signal is nearly up to the full power capability of the amplifier. At this signal level, the rectified output overcomes the delay bias and the gain of the preamplifier is reduced rapidly with increasing signal so that there is very little rise in output power above the threshold of gain control.

When a signal peak arrives that would normally overload the power amplifier, it is desireable that the gain of the ALC amplifier be reduced in a few milliseconds to a value where overloading of the power amplifier is overcome. After the signal peak passes, the gain should return to the normal value in about one-tenth second. These attack and release times are commonly used for voice communications. For this type of work, a dynamic range of at least 10 db is desirable. Input peaks as high as 20 db above the threshold of compression should not cause loss of control although some increase in distortion in the upper range of compression can be tolerated because peaks in this range are infrequent. Another limitation is that the preceding SSB generator must be capable of passing signals above full power output by the amount of compression desired. Since the signal level through the SSB generator should be maintained within a limited range, it is unlikely that more than 12 db ALC action will be useful. If the input signal varies more than this, a speech compressor should be used to limit the range of the signal fed into the SSB generator.

Figure 33 shows the effectiveness of the ALC in limiting the output signal to the capabilities of the power amplifier. An adjustment of the delay bias will place the threshold of compression at the desired power output. Figure 34 shows a simplified schematic of an ALC system. This ALC uses two variable gain am-

Figure 34
SIMPLIFIED SCHEMATIC OF AUTOMATIC LOAD CONTROL AMPLIFIER. OPERATING POINT OF ALC CIRCUIT MAY BE SET BY VARYING BLOCKING BIAS ON CATHODE OF 6X4 SIGNAL RECTIFIER



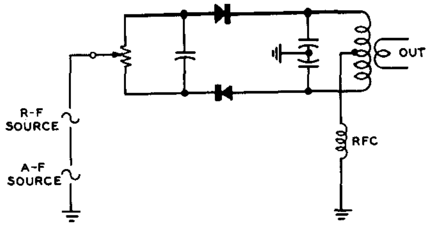


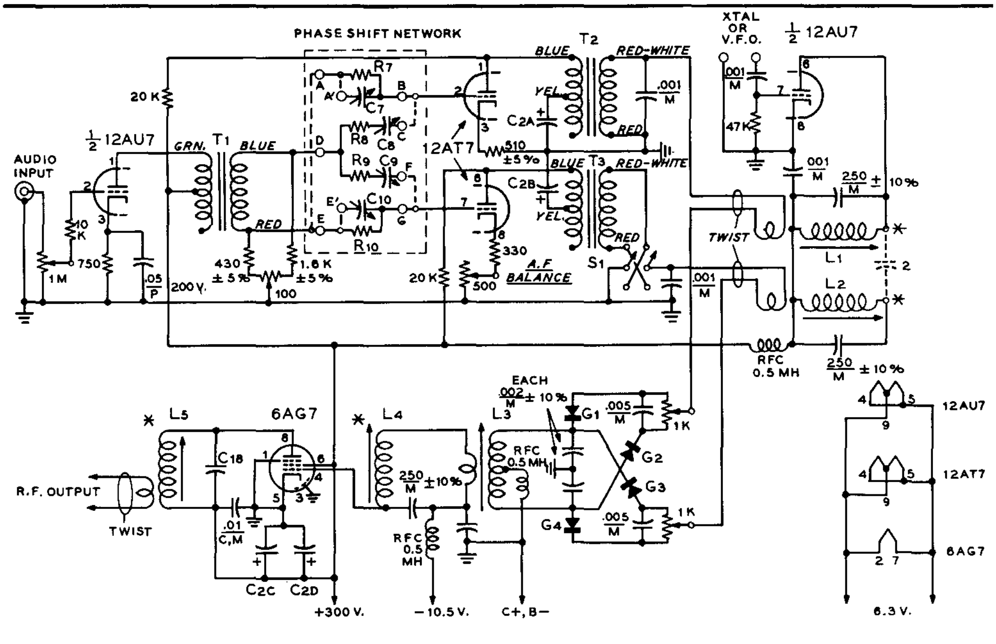
Figure 35
SSB JR. MODULATOR CIRCUIT
R-F and A-F sources are applied in series to balanced modulator.

plifier stages and the maximum overall gain is about 20 db. A meter is incorporated which is calibrated in db of compression. This is useful in adjusting the gain for the desired amount of load control. A capacity voltage divider is used to step down the r-f voltage at the plate of the amplifier tube to about 50 volts for the ALC rectifier. The output of the ALC rectifier passes through R-C networks to obtain the desired attack and release times and through r-f filter capacitors. The 3.3K resistor and 0.1 μfd. capacitor across the rectifier output stabilizes the gain around the ALC loop to prevent "motor-boating."

17-7 Sideband Exciters

Some of the most popular sideband exciters in use today are variations of the simple phasing circuit introduced in the November, 1950 issue of *General Electric Ham News.* Called the *SSB, Jr.,* this simple exciter is the basis for many of the phasing transmitters now in use. Employing only three tubes, the *SSB, Jr.* is a classic example of sideband generation reduced to its simplest form.

The SSB, Jr. This phasing exciter employs audio and r-f phasing circuits to produce a SSB signal at one spot frequency. The circuit of one of the balanced modulator stages is shown in figure 35. The audio signal and r-f source are applied in series to two germanium diodes serving as balanced modulators



C2A,B,C,D = EACH SECTION 20 µF, 450 V. ELECTROLYTIC
C7 = 2430 µµFD (.002 µFD MICA ±5% WITH 170-780 µµFD TRIMMER)
C8 = 4860 µµFD. (.0043 µFD MICA ±5% WITH 170-780 µµFD TRIMMER)
C9 = 1215 µµFD. (.001 µFD MICA ±5% WITH 50-380 µµFD TRIMMER)
C10 = 607.5 µµFD (500 µµFD MICA ±5% WITH 9-180 µµFD TRIMMER)
C18 = 350 µµFD 600 V. MICA ± 10% (250 µµFD AND 100 µµFD PARALLEL)
R7,R10 = 133,300 OHMS, 1/2 WATT ± 1%
R8,R9 = 100,000 OHMS, 1/2 WATT ± 1%
T1 = *STANCOR A-53C* TRANSFORMER.
T2,T3 = *UTC R-38A* TRANSFORMER.
S1 = DPDT TOGGLE SWITCH

G1,2,3,4 = 1N52 GERMANIUM DIODE OR EQUIVALENT
L1, L2 = 33 T. Nº 21 E. WIRE CLOSEWOUND ON *MILLEN Nº 69046* IRON CORE ADJUSTABLE SLUG COIL FORM. LINK OF 6 TURNS OF HOOKUP WIRE WOUND ON OPEN END.
L3 = 16 T. Nº 19 E. WIRE SPACED TO FILL *MILLEN Nº 69046* COIL FORM. TAP AT 8 TURNS. LINK OF 1 TURN AT CENTER.
L4 = SAME AS L1 EXCEPT NO LINK USED.
L5 = 28 T. OF Nº 19 E. WIRE. LINK ON END TO MATCH LOAD. (4 TURN LINK MATCHES 72 OHM LOAD)

* = MOUNTING END OF COILS.

Figure 36
SCHEMATIC, SSB, JR.

having a push-pull output circuit tuned to the r-f "carrier" frequency. The modulator drives a linear amplifier directly at the output frequency. The complete circuit of the exciter is shown in figure 36.

The first tube, a 12AU7, is a twin-triode serving as a speech amplifier and a crystal oscillator. The second tube is a 12AT7, acting as a twin channel audio amplifier following the phase-shift audio network. The linear amplifier stage is a 6AG7, capable of a peak power output of 5 watts.

Sideband switching is accomplished by the reversal of audio polarity in one of the audio channels (switch S_1), and provision is made for equalization of gain in the audio channels (R_{12}). This adjustment is necessary in order to achieve normal sideband cancellation, which may be of the order of 35 db or better. Phase-shift network adjustment may be achieved by adjusting potentiometer R_5. Stable modulator balance is achieved by the balance potentiometers R_{16} and R_{17} in conjunction with the germanium diodes.

The *SSB, Jr.* is designed for spot frequency operation. Note that when changing frequency L_1, L_2, L_3, L_4, and L_6 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. The principal effect of mistuning L_2, however, will be degraded sideband suppression.

Power requirements of the *SSB, Jr.* are 300 volts at 60 ma., and −10.5 volts at 1 ma. Under load the total plate current will rise to about 80 ma. at full level with a single tone input. With speech input, the total current will rise from the resting value of 60 ma. to about 70 ma., depending upon the voice waveform.

The "Ten-A" Exciter The *Model 10-A* phasing exciter produced by *Central Electronics, Inc.* is an advanced version of the *SSB, Jr.* incorporating extra features such as VFO control, voice operation, and multi-band operation. A simplified schematic of the *Model 10A* is shown in figure 37. The 12AX7 two stage speech amplifier excites a transformer coupled ½-12BH7 low impedance driver stage and a voice operated (VOX) relay system employing a 12AX7 and a 6AL5. A transformer coupled 12AT7 follows the audio phasing network, providing two audio channels having a 90-degree phase difference. A simple 90-degree r-f phase shift network in the plate circuit of the 9 Mc. crystal oscillator stage works into the matched, balanced modulator consisting of four 1N48 diodes.

The resulting 9 Mc. SSB signal may be converted to the desired operating frequency in a 6BA7 mixer stage. Eight volts of r-f from an external v-f-o injected on grid #1 of the 6BA7 is sufficient for good conversion efficiency and low distortion. The plate circuit of the 6BA7 is tuned to the sum or difference mixing frequency and the resulting signal is amplified in a 6AG7 linear amplifier stage. Two "tweet" traps are incorporated in the 6BA7 stage to reduce unwanted responses of the mixer which are apparent when the unit is operating in the 14 Mc. band. Band-changing is accomplished by changing coils L_8 and L_9 and the frequency of the external mixing signal. Maximum power output is of the order of 5 watts at any operating frequency.

A Simple 80 Meter Phasing Exciter A SSB exciter employing r-f and audio phasing circuits is shown in figure 38. Since the r-f phasing circuits are balanced only at one frequency of operation, the phasing exciter is necessarily a single frequency transmitter unless provisions are made to re-balance the phasing circuits every time a frequency shift is made. However for mobile operation, or spot frequency operation a relatively simple phasing exciter may be made to perform in a satisfactory manner.

A 12AU7 is employed as a Pierce crystal oscillator, operating directly on the chosen SSB frequency in the 80 meter band. The second section of this tube is used as an isolation stage, with a tuned plate circuit, L_1. The output of the oscillator stage is link coupled to a 90° r-f phase-shift network wherein the audio signal from the audio phasing network is combined with the r-f signals. Carrier balance is accomplished by adjustment of the two 1000 ohm potentiometers in the r-f phase network. The output of the r-f phasing network is coupled through L_2 to a single 6CL6 linear amplifier which delivers a 3 watt peak SSB signal on 80 meters.

A cascade 12AT7 and a single 6C4 comprise the speech amplifier used to drive the audio phase shift network. A small inter-stage transformer is used to provide the necessary 180° audio phase shift required by the network. The output of the audio phasing network is coupled to a 12AU7 dual cathode follower which provides the necessary low impedance circuit to match the r-f phasing network. A double-

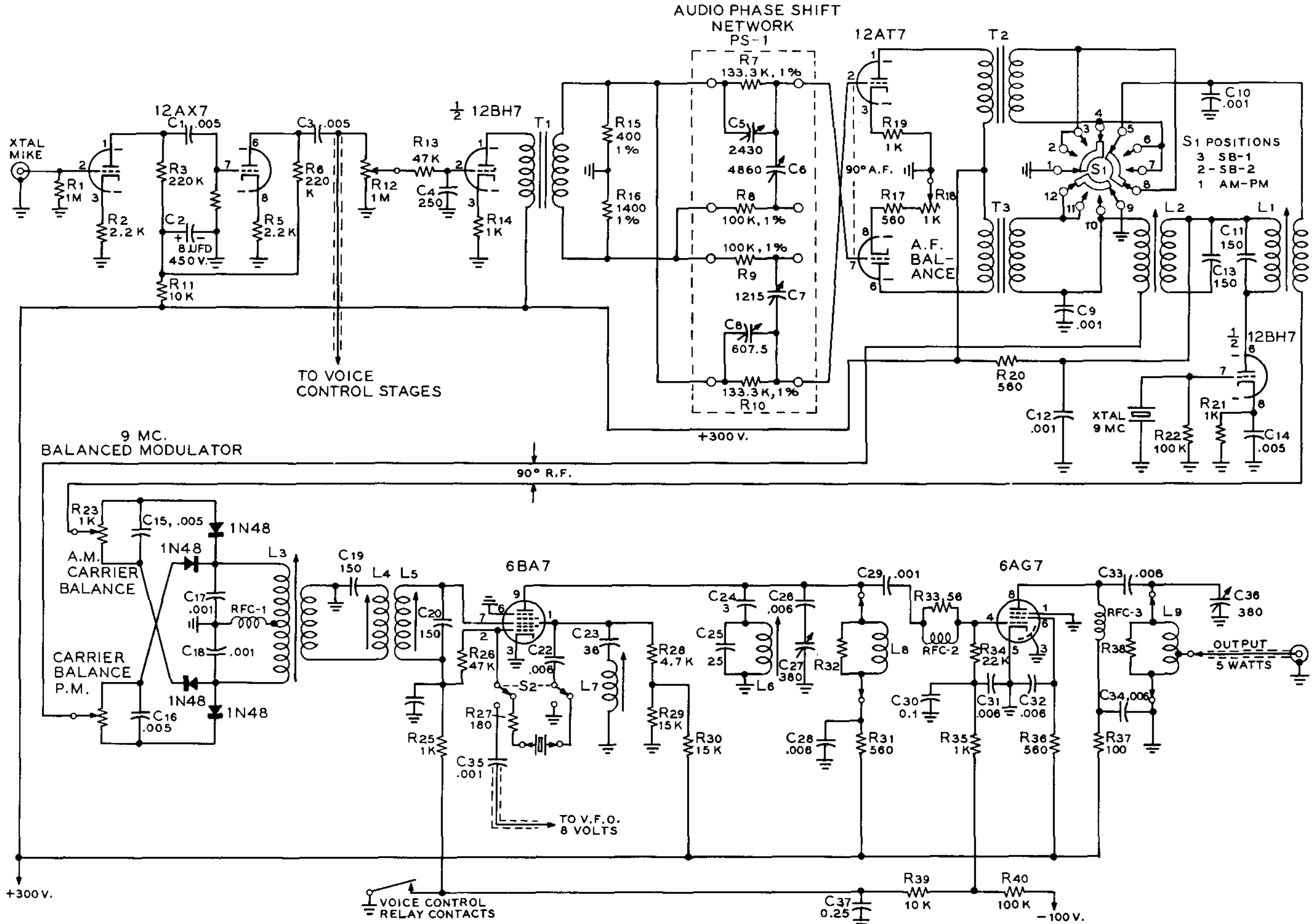


Figure 37
SIMPLIFIED SCHEMATIC OF "TEN-A" EXCITER

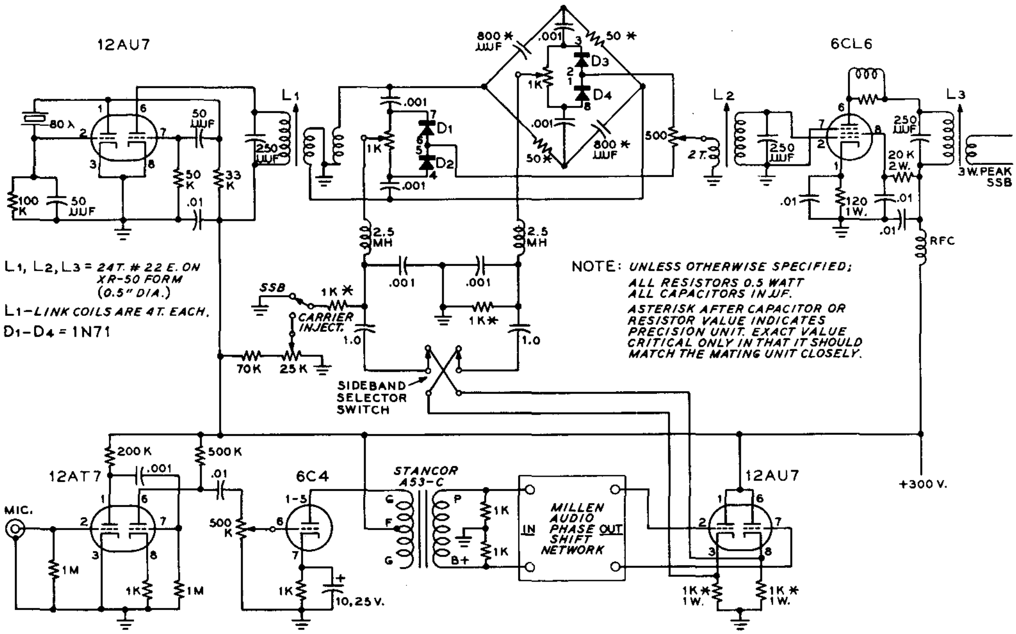


Figure 38
SIMPLE 3-WATT PHASING TYPE SSB EXCITER

pole double-throw switch in the output circuit of the cathode follower permits sideband selection.

A Filter-Type Exciter for 80 and 40 Meters A simple SSB filter-type exciter employing the Collins mechanical filter illustrates many of the basic principles of sideband generation. Such an exciter is shown in figure 39. The exciter is designed for operation in the 80 or 40 meter phone bands and delivers sufficient output to drive a class AB_1 tetrode such as the 2E26, 807, or 6146. A conversion crystal may be employed, or a separate conversion v-f-o can be used as indicated on the schematic illustration.

The exciter employs five tubes, exclusive of power supply. They are: 6U8 low frequency oscillator and r-f phase inverter, 6BA6 i-f amplifier, 6BA7 high frequency mixer, 6AG7 linear amplifier, and 12AU7 speech amplifier and cathode follower. The heart of the exciter is the balanced modulator employing two 1N81 germanium diodes and the 455 kc., 3500-cycle bandwidth mechanical filter. The input and output circuits of the filter are resonated to 455 kc. by means of small padding capacitors.

A series-tuned Clapp oscillator covers the range of 452 kc. — 457 kc. permitting the carrier frequency to be adjusted to the "20 decibel" points on the response curve of the filter, as shown in figure 40. Proper r-f signal balance to the diode modulator may be obtained by adjustment of the padding capacitor in the cathode circuit of the triode section of the 6U8 r-f tube. Carrier balance is set by means of a 50K potentiometer placed across the balanced modulator.

One half of a 12AU7 serves as a speech amplifier delivering sufficient output from a high level crystal microphone to drive the second half of the tube as a low impedance cathode follower, which is coupled to the balanced modulator. The two 1N81 diodes act as an electronic switch, impressing a double sideband, suppressed-carrier signal upon the mechanical filter. By the proper choice of frequency of the beating oscillator, the unwanted sideband may be made to fall outside the passband of the mechanical filter. Thus a single sideband suppressed-carrier signal appears at the output of the filter. The 455 kc. SSB signal is amplified by a 6BA6 pentode stage, and is then converted to a frequency in the 80 meter or 40 meter band by a 6BA7 mixer stage. Either a crystal or an external v-f-o may be used for the mixing signal.

To reduce spurious signals, a double tuned

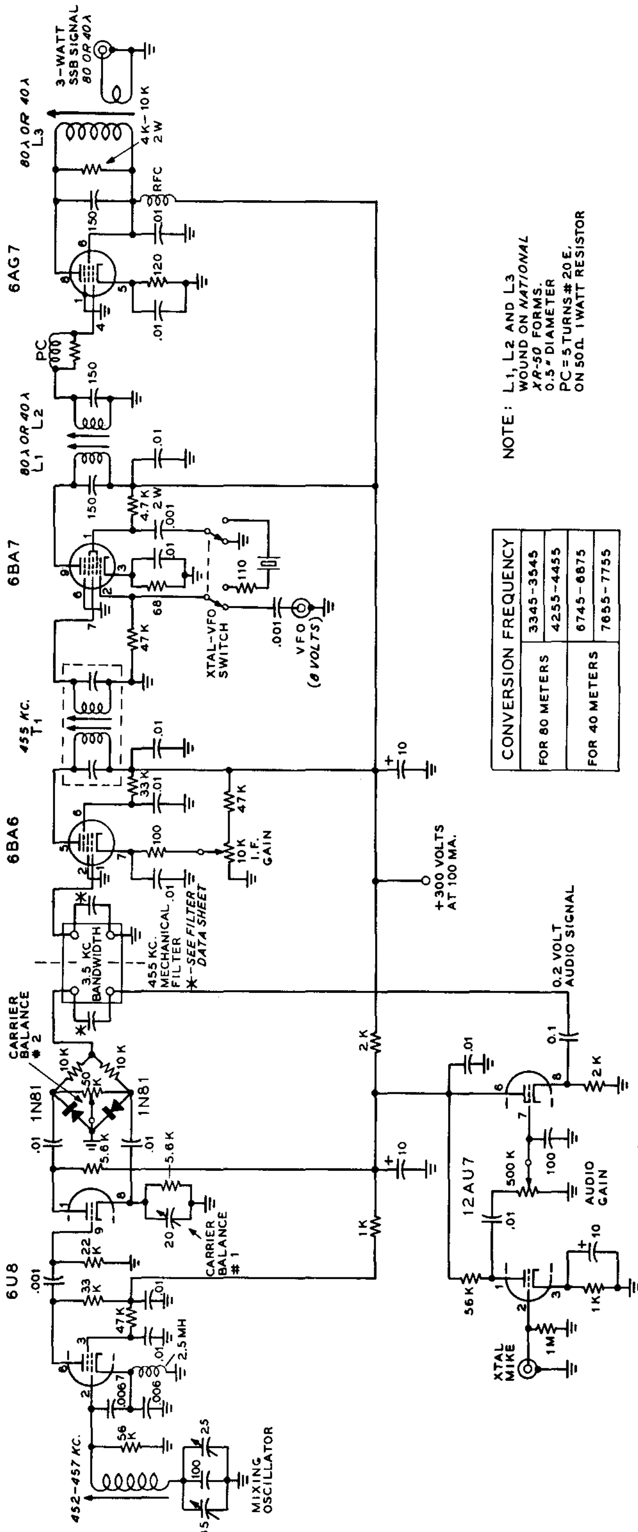


Figure 39
SCHEMATIC, FILTER-TYPE SSB EXCITER FOR 80 OR 40 METER OPERATION

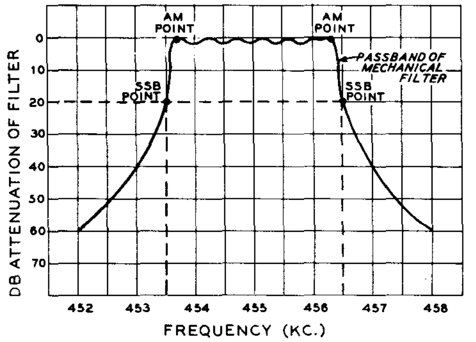


Figure 40
THE "TWENTY DB" CARRIER POINTS ON THE FILTER CURVE
The beating oscillator should be adjusted so that its frequency corresponds to the 20 db attenuation points of the mechanical filter passband. The carrier of the SSB signal is thus attenuated 20 db in addition to the inherent carrier attenuation of the balanced mixer. A total carrier attenuation of 50 db is achieved. Unwanted sideband rejection is of the same order.

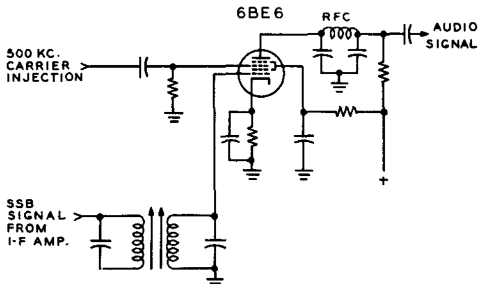


Figure 41
THE PRODUCT DETECTOR
The above configuration resembles pentagrid converter circuit.

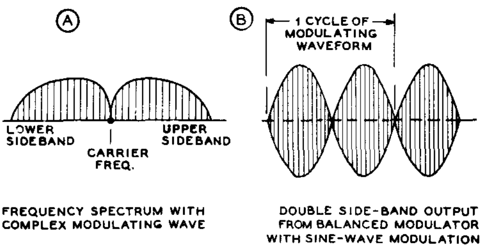


Figure 42
DOUBLE-SIDEBAND SUPPRESSED-CARRIER SIGNAL
The envelope shown at B also is obtained on the oscilloscope when two audio frequencies of the same amplitude are fed to the input of a single-sideband transmitter.

transformer is placed between the mixer stage and the 6AG7 output stage. A maximum signal of 3 watts may be obtained from the 6AG7 linear amplifier.

Selection of the upper or lower sideband is accomplished by tuning the 6U8 beating oscillator across the passband of the mechanical filter, as shown in figure 40. If the 80 meter conversion oscillator is placed on the low frequency side of the SSB signal, placing the 6U8 beating oscillator on the low frequency side of the passband of the mechanical filter will produce the upper sideband on 80 meters. When the beating oscillator is placed on the high frequency side of the passband of the mechanical filter the lower sideband will be generated on 80 meters. If the 80 meter conversion oscillator is placed on the high frequency side of the SSB signal, the sidebands will be reversed from the above. The variable oscillator should be set at approximately the 20 db suppression point of the passband of the mechanical filter for best operation, as shown in figure 40. If the oscillator is closer in frequency to the filter passband than this, carrier rejection will suffer. If the oscillator is moved farther away in frequency from the passband, the lower voice frequencies will be attenuated, and the SSB signal will sound high-pitched and tinny. A little practice in setting the frequency of the beating oscillator while monitoring the 80 meter SSB signal in the station receiver will quickly acquaint the operator with the proper frequency setting of the beating oscillator control for transmission of either sideband.

If desired, an amplitude modulated signal with full carrier and one sideband may be transmitted by placing the 6U8 low frequency oscillator just inside either edge of the passband of the filter (designated "AM point", figure 40).

After the 6U8 oscillator is operating over the proper frequency range it should be possible to tune the beating oscillator tuning capacitor across the passband of the mechanical filter and obtain a reading on the S-meter of a receiver tuned to the filter frequency and coupled to the input grid of the 6BA6 i-f amplifier tube. The two carrier balance controls of the 6U8 phase inverter section should be adjusted for a null reading of the S-meter when the oscillator is placed in the center of the filter passband. The 6BA6 stage is now checked for operation, and transformed T_1 aligned to the carrier frequency. It may be necessary to unbalance temporarily potentiometer #2 of the 6U8 phase inverter in order to obtain a sufficiently strong signal for proper alignment of T_1.

A conversion crystal is next plugged in the 6BA7 conversion oscillator circuit, and the operation of the oscillator is checked by monitoring the crystal frequency with a nearby receiver. The SSB "carrier" produced by the unbalance of potentiometer #2 should be heard at the proper sideband frequency in either the 80 meter or 40 meter band. The coupled circuit between the 6BA7 and the 6AG7 is resonated for maximum carrier voltage at the grid of the amplifier stage. Care should be taken that this circuit is tuned to the sideband frequency and not to the frequency of the conversion oscillator. Finally, the 6AG7 stage is tuned for maximum output. When these adjustments have been completed, the 455 kc. beating oscillator should be moved just out of the passband of the mechanical filter. The 80 meter "carrier" will disappear. If it does not, there is either energy leaking around the filter, or the amplifier stages are oscillating. Careful attention to shielding (and neutralization) should cure this difficulty.

Audio excitation is now applied to the exciter, and the S-meter of the receiver should kick up with speech, but the audio output of the receiver should be unintelligible. As the frequency of the beating oscillator is adjusted so as to bring the oscillator frequency within the passband of the mechanical filter the modulation should become intelligible. A single sideband a.m. signal is now being generated. The BFO of the receiver should now be turned on, and the beating oscillator of the exciter moved out of the filter passband. When the receiver is correctly tuned, clean, crisp speech should be heard. The oscillator should be set at one of the "20 decibel" points of the filter curve, as shown in figure 40 and all adjustments trimmed for maximum carrier suppression.

17-8 Reception of Single Sideband Signals

Single-sideband signals may be received, after a certain degree of practice in the technique, in a quite adequate and satisfactory manner with a good communications receiver. However, the receiver must have quite good frequency stability both in the high-frequency oscillator and in the beat oscillator. For this reason, receivers which use a crystal-controlled first oscillator are likely to offer a greater degree of satisfaction than the more common type which uses a self-controlled oscillator.

Beat oscillator stability in most receivers is usually quite adequate, but many receivers do not have a sufficient amplitude of beat oscillator injection to allow reception of strong SSB signals without distortion. In such receivers it is necessary either to increase the amount of beat-oscillator injection into the diode detector, or the manual gain control of the receiver must be turned down quite low.

The tuning procedure for SSB signals is as follows: The SSB signals may first be located by tuning over the band with receiver set for the reception of c-w.; that is, with the manual gain at a moderate level and with the beat oscillator operating. By tuning in this manner SSB signals may be *located* when they are far below the amplitude of conventional AM signals on the frequency band. Then after a signal has been located, the beat oscillator should be turned off and the receiver put on a.v.c. Following this the receiver should be tuned for maximum swing of the S meter with modulation of the SSB signal. It will not be possible to understand the SSB signal at this time, but the receiver may be tuned for maximum deflection. Then the receiver is put back on manual gain control, the beat oscillator is turned on again, the manual gain is turned down until the background noise level is quite low, and the *beat oscillator* control is varied until the signal sounds natural.

The procedure in the preceding paragraph may sound involved, but actually all the steps except the last one can be done in a moment. However, the last step is the one which will require some practice. In the first place, it is not known in advance whether the upper or lower sideband is being transmitted. So it will be best to start tuning the beat oscillator from one side of the pass band of the receiver to the other, rather than starting with the beat oscillator near the center of the pass band as is normal for c-w reception.

With the beat oscillator on the wrong side of the sideband, the speech will sound inverted; that is to say that low-frequency modulation tones will have a high pitch and high-frequency modulation tones will have a low pitch—and the speech will be quite unintelligible. With the beat oscillator on the correct side of the sideband but too far from the correct position, the speech will have some intelligibility but the voice will sound quite high pitched. Then as the correct setting for the beat oscillator is approached the voice will begin to sound natural but will have a background growl on each syllable. At the correct frequency for the beat oscillator the speech will clear completely and the voice will have a clean, crisp quality. It should also be mentioned that there is a narrow region of tuning of the beat oscillator a small distance on the wrong side of the sideband where the voice will sound quite bassy and difficult to understand.

With a little experience it will be possible to identify the sound associated with improper settings of the beat-oscillator control so that corrections in the setting of the control can be made. Note that the main tuning control of the receiver is not changed after the sideband once is tuned into the pass band of the receiver. All the fine tuning should be done with the beat oscillator control. Also, it is very important that the r-f gain control be turned to quite a low level during the tuning process. Then after the signal has been tuned properly the r-f gain may be increased for good signal level, or until the point is reached where best oscillator injection becomes insufficient and the signal begins to distort.

Single-Sideband Receivers and Adapters Greatly simplified tuning, coupled with strong attenuation of undesired signals, can be obtained through the use of a single-sideband receiver or receiver adapter. The exalted carrier principle usually is employed in such receivers, with a phase-sensitive system sometimes included for locking the local oscillator to the frequency of the carrier of the incoming signal. In order for the locking system to operate, some carrier must be transmitted along with the SSB signal. Such receivers and adapters include a means for selecting the upper or lower sideband by the simple operation of a switch. For the reception of a single-sideband signal the switch obviously must be placed in the correct position. But for the reception of a conventional AM or phase-modulated signal, either sideband may be selected, allowing the sideband with the least interference to be used.

The Product Detector An unusually satisfactory form of demodulator for SSB service is the *product detector,* shown in one form in figure 41. This circuit is preferred since it reduces intermodulation products and does not require a large local carrier voltage, as contrasted to the more common diode envelope detector. This product detector operates much in the same manner as

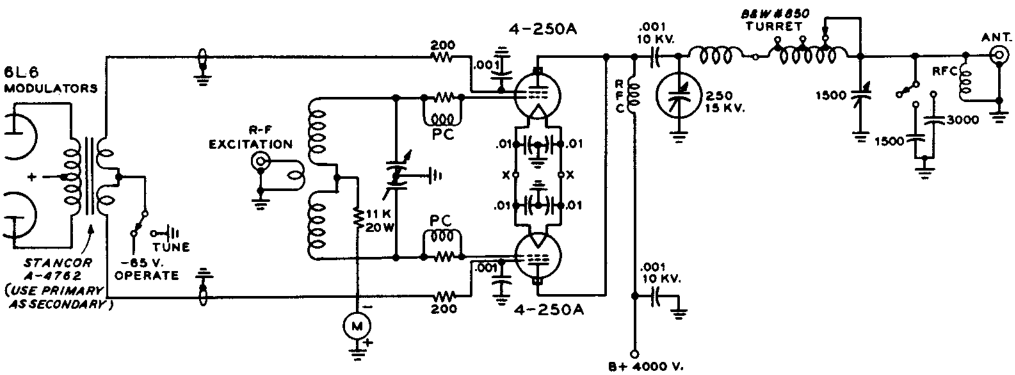


Figure 43
HIGH-LEVEL DSB BALANCED MODULATOR

a multi-grid mixer tube. The SSB signal is applied to the control grid of the tube and the locally generated carrier is impressed upon the other control grid. The desired audio output signal is recovered across the plate resistance of the demodulator tube. Since the cathode current of the tube is controlled by the simultaneous action of the two grids, the current will contain frequencies equal to the sum and difference between the sideband signal and the carrier. Other frequencies are suppressed by the low-pass r-f filter in the plate circuit of the stage, while the audio frequency is recovered from the i-f sideband signal.

17-9 Double Sideband Transmission

Many systems of intelligence transmission lie in the region between amplitude modulation on the one hand and single sideband suppressed-carrier transmission on the other hand. One system of interest to the amateur is the *Synchronous Communications System,* popularly known as "double sideband" (DSB) transmission, wherein a suppressed-carrier double sideband signal is transmitted (figure 42). Reception of such a signal is possible by utilizing a local oscillator phase-control system which derives carrier phase information from the sidebands alone and does not require the use of any pilot carrier.

The DSB Transmitter A balanced modulator of the type shown in figure 8 may be employed to create a DSB signal. For higher operating levels, a pair of class-C type tetrode amplifier tubes may be screen modulated by a push-pull audio system and excited from a push-pull r-f source. The plates of such a modulator are connected in parallel to the tank circuit, as shown in figure 43. This DSB modulator is capable of 1-kilowatt peak power output at a plate potential of 4000 volts. The circuit is self-neutralizing and the tune-up process is much the same as with any other class-C amplifier stage. As in the case of SSB, the DSB signal may also be generated at a low level and amplified in linear stages following the modulator.

Synchronous Detection A DSB signal may be received with difficulty on a conventional receiver, and one of the two sidebands may easily be received on a single sideband receiver. For best reception, however, a phase-locked local oscillator and a synchronous detector should be employed. This operation may be performed either at the frequency of reception or at a convenient intermediate frequency. A block dia-

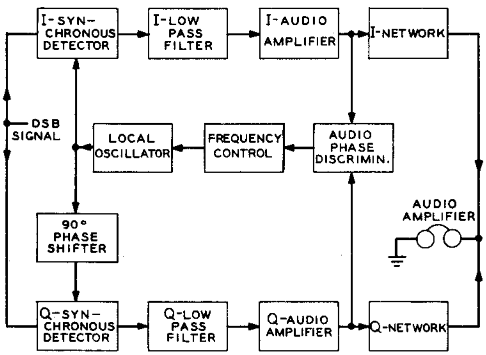


Figure 44
BLOCK DIAGRAM OF DSB RECEIVING ADAPTER

gram of a DSB synchronous receiver is shown in figure 44. The DSB signal is applied to two detectors having their local oscillator conversion voltages in phase quadrature to each other so that the audio contributions of the upper and lower sidebands reinforce one another. The *in-phase* oscillator voltage is adjusted to have the same phase as the suppressed carrier of the transmitted signal. The I-amplifier audio output, therefore, will contain the demodulated audio signal, while the Q-amplifier (supplied with *quadrature* oscillator voltage) will produce no output due to the quadrature null. Any frequency change of the local oscillator will produce some audio output in the Q-amplifier, while the I-amplifier is relatively unaffected. The Q-amplifier audio will have the same polarity as the I-channel audio for one direction of oscillator drift, and opposite polarity for oscillator drift in the opposite direction. The Q-amplifier signal level is proportional to the magnitude of the local oscillator phase angle error (the oscillator drift) for small errors. By combining the I-signal and the Q-signal in the audio phase discriminator a d-c control voltage is developed which automatically corrects for local oscillator phase errors. The reactance tube therefore locks the local oscillator to the correct phase. Phase control information is derived entirely from the sideband component of the signal and the carrier (if present) is not employed. Phase control ceases with no modulation of the signal and is reestablished with the reappearance of modulation.

Interference Rejection Interference falling within the passband of the receiver can be reduced by proper combination of the I- and Q- audio signals. Under phase lock conditions, the I-signal is composed of the audio signal plus the undesired interference, whereas the Q-signal contains only the interference component. Phase cancellation obtained by combining the two signals will reduce the interference while still adding the desired information contained in both side-bands. The degree of interference rejection is dependent upon the ratio of interference falling upon the two sidebands of the received signal and upon the basic design of the audio networks. A schematic and description of a complete DSB receiving adapter is shown in the June, 1957 issue of *CQ* magazine.